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Can ensemble condition in a hall be improved and measured?

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PROGRAM OF

Acoustical Society of America Acoustical Society of Japan

Second Joint Meeting

Sheraton-Waikiki Hotel • Honolulu, Hawaii • 14–18 November 1988

MONDAY AFTERNOON, 14 NOVEMBER 1988 KAUAI ROOM, 1:30 TO 3:30 P.M.

Session TU1. Tutorial on Digital Signal Processing

Hiroya Fujisaki, Chairman
Department of Electronic Engineering, Faculty of Engineering, University of Tokyo, 7-3-1 Hongo, Bunkyo-ku, Tokyo, 113 Japan

An introduction to digital signal processing. Alan V. Oppenheim (Room 36-615, Department of Electrical Engineering and Computer Science, Massachusetts Institute of Technology, Cambridge, MA 02139)

The basic techniques of digital signal processing play an important role in the processing of a wide variety of acoustic data. This tutorial presentation is intended to introduce the basic techniques of digital signal processing to those not presently familiar with them, with an emphasis on time-invariant nonadaptive techniques. The presentation will be in three parts. The first will consider sampling issues and discrete-time processing of continuous-time signals. The second part will focus on issues of digital filtering. Specifically, nonrecursive (moving average) filters and recursive (autoregressive) filters will be discussed along with difference equation and block diagram representations. In the third part, signal processing and spectral analysis based on the discrete Fourier transform (DFT) will be discussed, along with an introduction to algorithms for computation of the DFT. A brief discussion of some high-resolution spectral analysis methods may also be included.

MONDAY AFTERNOON, 14 NOVEMBER 1988 KAUAI ROOM, 4:00 TO 6:00 P.M.

Session TU2. Tutorial on Adaptive Signal Processing

John C. Burgess, Chairman
Department of Mechanical Engineering, University of Hawaii, 2540 Dole Street, Honolulu, Hawaii 96822

Adaptive signal processing in acoustics. M. Mohan Sondhi (Room 2D-536, AT&T Bell Laboratories, Murray Hill, NJ 07974)

In this tutorial the basic ideas about adaptive signal processing will be introduced in the context of system identification, inverse modeling, and control. The gradient algorithm for minimizing mean-squared error will be derived. Convergence properties of this algorithm will be discussed in some detail for the case when the model is nonrecursive. Several other model structures (e.g., recursive and lattice) and algorithms (e.g., block and recursive least squares) will be briefly touched upon. Finally, several applications of adaptive processing will be discussed. These include echo cancellation in rooms and on telephone circuits, active control of noise and reverberation, adaptive microphone arrays, etc.

Session A. Opening Plenary Session

W. Dixon Ward, *President*
Acoustical Society of America

Masaru Koyasu, *President*
Acoustical Society of Japan

Welcoming remarks in honor of the Second Joint Meeting of the Acoustical Societies of America and Japan

TUESDAY MORNING, 15 NOVEMBER 1988 HONOLULU/KAHUKU ROOM, 8:30 TO 9:30 A.M.

Session B. Architectural Acoustics I: Vern O. Knudsen Lecture

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Hughes Aircraft Company
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Minoru Nagata, Cochairman
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Tokyo, 160 Japan

Chairman's Introduction—8:30

Invited Paper

8:35

B1. The inner universe: A multidisciplinary approach to the acoustics of concert halls. Yoichi Ando (Faculty of Engineering, Kobe University, Rokkodai, Nada-ku, Kobe, 657 Japan)

After introducing dimensional factors of the acoustic system in a concert hall, the capability of calculating acoustical quality at any seat based upon the theory of preference is presented. The effort to describe important qualities of sound in terms of the processes of the auditory pathways and brain has been brought to bear on the problem. If enough were known about how the auditory and the central nervous systems modify the nerve impulses from the cochlea, the design of concert halls could proceed according to guidelines derived from knowledge of these processes. An attempt to analyze this is made through study of the auditory-evoked potentials over the left and right human cerebral hemispheres. In addition, results of recording the slow vertex responses (SVR), which were obtained by adjusting the temporal and spatial physical factors, indicated that the information related to subjective preference appeared in the latency components. The longest-latency responses were observed at the most-preferred condition or at the most subjectively diffuse condition. The correlation between brain activity and subjective preference seems to indicate that subjective preference can be traced back to a primitive response in the "inner universe." Thus, such a theory with temporal and spatial factors may be generalized in designing physical environments for every human activity.

Session C. Bioresponse to Vibration I: Hand–arm Vibration

Anthony J. Brammer, Cochairman
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Akira Okada, Cochairman
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Kanazawa University
13-1 Takaramachi
Kanazawa, 920 Japan

Invited Papers

8:30

C1. Hand–arm vibration: New perspectives. Robert L. Brubaker (Department of Health Care and Epidemiology, University of British Columbia, Vancouver, British Columbia V6T 1W5, Canada)

Signs and symptoms of the hand–arm vibration syndrome (HAVS) will be described followed by recent research developments. Much attention has been focused on elucidating the pathophysiological basis of vibration-induced injury to vascular, neural, muscular, and skeletal tissues in the hand and arm. New information suggests that chronic neural damage is less reversible after long-term cessation of vibration exposure than vascular damage. If confirmed, current hand–arm vibration standards (based mainly on prevention of onset of finger blanching symptoms) should be re-evaluated to also preclude onset of chronic neurologic deficits. Progress has been made in designing and validating new objective measures of peripheral somatosensory dysfunction, as well as improving and testing existing methods for assessing neural and vascular pathology. The Stockholm Workshop symptom scale (revised Taylor–Pelmear scale) for assessing the vascular component of HAVS will be described as well as a suggested new scale for evaluating neural damage.

8:50

C2. System for monitoring peripheral circulation in the diagnosis of vibration hazards. Seiichi Nohara and Hideki Nakamura (Department of Public Health, School of Medicine, Kanazawa University, 13-1 Takaramachi, Kanazawa, 920 Japan)

Peripheral circulatory disorders such as Raynaud's phenomenon are thought to be basal symptoms of vibration hazards. The elucidation of their pathogenesis and their diagnosis between attacks is regarded as important. A system for monitoring the body surface circulation noninvasively and continuously has been developed, which enables the quantitative dynamic assessment of skin blood flow. This apparatus is based on the thermal diffusion method and uses a probe incorporating a Peltier stack. After fundamental studies on this system, the peripheral circulatory functions of workers using vibrating tools were studied. The workers were divided into three groups: group A, without any symptoms; group B, with numbness and/or pain but without Raynaud's phenomenon; and group C, with Raynaud's phenomenon. The finger blood flow (FBF) of group C at rest was significantly lower than that of group A ($p<0.01$). In a handgrip test and with exposure to local vibration (60 Hz, 50 m/s^2), the FBF of group A significantly changed compared to its value at rest, but that of group C showed no significant change.

9:10

C3. Vibrotactile perception thresholds in hands occupationally exposed to vibration: Interpretation of sensorineural loss. J. E. Piercy, A. J. Brammer (Division of Physics, National Research Council, Ottawa, Ontario K1A 0R6, Canada), H. Nakamura, S. Nohara (Department of Public Health, School of Medicine, Kanazawa University, Kanazawa, 920 Japan), P. L. Auger (Département de santé communautaire du Centre Hopitalier de l'Université Laval, Ste-Foy, Québec G1V 2K8, Canada), and A. T. Haines (Department of Clinical Epidemiology and Biostatistics, McMaster University, Hamilton, Ontario L8N 3Z5, Canada)

The tactile performance of the hand is now known to be critically dependent on neural activity in one population of slow-adapting (SAI) and two populations of fast-adapting (FAI, FAII) mechanoreceptors. A technique has recently been developed to establish the sensitivity of these receptor populations at the fingertip from vibrational perception thresholds determined psychophysically with sinusoidal stimulation [A. J. Brammer *et al.*, J. Hand Surg. **12A**, 870–875 (1987)]. A comparison of data from 98 vibration-exposed and 34 normal hands, all screened to exclude confounding factors, has revealed two patterns of abnormal threshold elevation. The first, involving similarly elevated thresholds at all frequencies (2–200 Hz), is indicative of sensorineural losses of similar magnitude in each type of receptor-nerve system, and so is suggestive of peripheral nerve degeneration. The second pattern involves elevated thresholds at frequencies mediated only by one or sometimes two receptor types (commonly SAI and/or FAII). A mechanoreceptor-specific mechanism is unlikely to occur within a nerve trunk, and so this frequency-dependent pattern is suggestive of selective damage to the nerve endings.

9:30

C4. A study on the setting position of vibration pickups for the measurement of hand–arm transmitted vibration. Yasuo Tokita,[a] Atushi Oda (Kobayasi Institute of Physical Research, Kokubunji, 185 Japan), and Tunenobu Ohkuma (Rion Co., Ltd., Kokubunji, 185 Japan)

In order to reduce hand–arm vibration disease, it is important to know the amount of vibration transmitted from a hand-held vibration tool to the hand–arm system. There are many problems in determining the amount of exposure in practice. For example, (1) the handle vibration does not always correspond to the vibration disease. (2) If the vibration pickup is set between the handle and palm, this system will not be usable for a long time, because the vibration pickup will become an obstacle for the worker. In this study, L_{eq8} (8 h of equivalent hand–arm vibration level according to ISO 5349) is used to evaluate the transmitted vibration, and the back side of the hand is recommended for the placement of the vibration pickup to measure the amount of transmitted vibration without disturbing the worker during the long-time use of the measuring device. The reasons for this proposal are indicated by the results of field measurements of actual hand-held vibration tools (chain saw, breaker, etc.) and by the analytical results of the model experiment. [Work supported by Ministry of Labor of Japan.] [a] Presently at Aircraft Nuisance Research Center, Hanedakuko, Ohta-ku, Tokyo, 144 Japan.

Contributed Papers

9:50

C5. Errors in the measurement of power tool handle vibration from excess accelerometer weight. S. E. Keith (Department of Mechanical and Aeronautical Engineering, Carleton University, Ottawa, Ontario K1S 5B6, Canada) and A. J. Brammer (Division of Physics, National Research Council, Montreal Road, Ottawa, Ontario K1A 0R6, Canada)

ISO 7505 (Forest Machinery—Chain Saws—Measurement of hand-transmitted vibration. International Oragnization for Standardization, Geneva, 1986) specifies for the measurement of frequency-weighted handle acceleration a maximum permissible accelerometer weight of 50 g, which is comparable to that of most chain saw handles. To explore the magnitude of errors so introduced, a series of difference measurements has been conducted with a variable mass accelerometer mount. With this device the effective accelerometer weight can be cycled between 6 and 25, 50, or 100 g, in a time in which other parameters affecting handle vibration during saw operation with no load (viz: engine speed and hand grip) usually vary little. Reductions in frequency-weighted acceleration from the minimum resolvable ($\pm 5\%$) to 50% were recorded when changing accelerometer weight from 6 to 50 g, depending on handle design and material. Investigation of the vibration spectra (6–1250 Hz) and transmissibility (30–700 Hz) of one handle suggests that the data are consistent with a localized mass loading, introduced by the accelerometer, reducing the amplitude of one or more flexural modes of handle vibration.

10:02

C6. Materials performance evaluation for hand–arm vibration isolation. Gary A. Hampel (Loss Prevention Department, Liberty Mutual Insurance Company, Research Center, 71 Frankland Road, Hopkinton, MA 01748)

Materials performance affecting the level of surface vibration ordinarily in direct contact with the hand under specified conditions will be reported. The intent of this study was to develop a method by which elastic materials can be evaluated for isolation properties relating to test surface vibration using nonrigid loading. The hand and arm, as a complex system, are used in place of the more traditional and simpler rigid mass load. This research and the scope of these findings report performance as a more realistic means for determining isolation, i.e., vibration reduction caused by an intervening material. The findings reported use this method to identify materials that exhibit isolation and display this positive effect, if it occurs, by graphic means over the frequency range of 1 to 800 Hz. The Z-axis (basicentric) hand–arm apparent mass, evaluated while grasping a 3.81-cm-diam handle with 4.5-Newton total grip force accelerated at 4 m/s^2, will be used as the reference load to determine material isolation effectiveness, reported as transmissibility.

TUESDAY MORNING, 15 NOVEMBER 1988 — WAIANAE ROOM, 8:30 TO 11:06 A.M.

Session D. Engineering Acoustics I: Theory, Practice, and Materials in Engineering Acoustics

Sung H. Ko, Cochairman
Naval Underwater Systems Center
New London, Connecticut 06320

Hikaru Date, Cochairman
Department of Information Engineering
Yamagata University
Yonezawa, 992 Japan

Contributed Papers

8:30

D1. Signal pressure received by a hydrophone placed on a plate backed with a compliant baffle. Sung H. Ko and Howard H. Schloemer (Naval Underwater Systems Center, New London, CT 06320)

A theoretical model was developed to evaluate the signal pressure received by a hydrophone placed in front of a plate backed with a compliant baffle layer. The compliant baffle layer between the plate and the semi-infinite fluid medium is designed for reducing pressure fluctuations from nonsignal directions. Because of its acoustic softness, the signal received by the hydrophone without a plate would be degraded. Therefore, it is desirable to improve the signal reception by covering the baffle layer with a hard plate. The baffle layer considered here is the compliant-tube array, modeled by Junger [J. Acoust. Soc. Am. **78**, 1010 (1985)], to represent a homogeneous (dispersive) fluid layer. Effects of various parameters such as the angle of incidence, the aspect ratio of the compliant tube, the distance between tubes, and the damping of the tube on the received signal pressure are presented. Calculations made for the nondispersive fluid layer are compared with those made for the dispersive fluid layer.

8:42

D2. Optimization of a low-frequency transmitting array. Dominique Lalisse and Didier Boucher (GERDSM, DCAN Toulon, DCN, Le Brusc, 83140 Six-Fours-Les-Plages, France)

A low-frequency plane array is studied in water in a wide frequency band around the resonance frequency of the transducers. The array under study is made of eight length-expander vibrators (two columns of four transducers) with a circular radiating face in a rigid box of limited dimensions. The radiating impedance matrices are calculated by an integral equation method [C. Audoly, J. Acoust. Soc. Am. Suppl. 1 **83**, S20 (1988)] and projectors are modeled with a classical electromechanical equivalent circuit. Due to the effects of acoustic interactions, no terms in the matrices are found to be negligible. Mechanical and electrical constraints on the transducers are identified and computed. The array is studied under three conditions: identical voltage driving, identical headmass velocity distribution, and acoustic power optimization. The results confirm that acoustic interactions have important and drastic effects around the resonance. The study of acoustic power optimization makes it possible to discuss the opportunity of using velocity control and electromechanical feedback devices in low-frequency sonar projector arrays.

8:54

D3. Comparison of theoretical and experimental form functions for a viscoelastic coated infinite shell. Gary W. Caille, Jacek Jarzynski, and Peter H. Rogers (George W. Woodruff School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

The experimentally observed backscattered farfield form functions (normal incidence) for a simulated infinite cylindrical shell coated with (a) closed-cell nitrile rubber and (b) corprene are compared with the theoretically derived form functions. The shell is 304 stainless steel with a radius ratio of 0.97 and a length greater than 8 ft so as to avoid end effects. Air is the inner fluid. The coatings have specific acoustic impedances less than water and the nitrile coating is highly attenuating. The experiment was conducted for an approximate *ka* range of 1.5–15. The viscoelastic constants for the coatings were determined from the experimentally measured Young's and plane-wave moduli. The experimental procedure for measuring the form function was validated by the excellent agreement of the observed form function with the theoretical form function for the shell only. Significant observations are that (1) these coatings increase the form functions relative to the shell at low *ka* values; (2) the coatings reduce or eliminate the resonances observed in the shell-only baseline; and (3) the nitrile-coated shell is an excellent simulation of an ideal pressure release cylinder. [Work supported by Office of Naval Research, Code 11250A.]

9:06

D4. Radiation from an array of simple sources. Adnan Akay (Department of Mechanical Engineering, Wayne State University, Detroit, MI 48202)

General closed-form expressions for the farfield intensity and power radiated by finite arrays are derived for a class of linear arrays made up of simple point sources. The closed-form solutions are made possible by a special trigonometric relationship that simplifies the intensity expression. Earlier work on out-of-phase sources [J. Acoust. Soc. Am. Suppl. 1 **79**, S30 (1986)] has been extended to include in-phase sources with axes either transverse or coincident with the axis of the array. [Work sponsored by NSF.]

9:18

D5. The increase of transducer directivity using diffractive attachments. Wieslaw R. Woszczyk (Graduate Sound Recording Program, Faculty of Music, McGill University, 555 Sherbrooke Street West, Montreal, Quebec H3A 1E3, Canada)

Increased directivity is sought for pressure microphones at frequencies above 1 kHz in order to accomplish useful directional selectivity and better signal-to-noise ratio in the pickup of direct transient information contained within reverberant sound fields. The goal is to improve the "reach" of pressure microphones for direct transient sound under reverberant conditions in rooms while maintaining only single-point sampling of the sound field. Smooth off-axis frequency responses and undistorted time-domain responses of microphones are desired to transcribe with fidelity the complex sound field at the point of pickup. Diffractive and absorptive attachments are installed on microphones to modify their frequency and directional responses without distorting time-domain responses. Detailed directional frequency responses and transient responses are measured at every 5 and 15 deg of angular sound incidence using impulse techniques to verify the effect of the attachments. These responses are compared to those of high-quality microphones used for the recording of music and speech. [Work supported by SSHRC.]

9:30

D6. Effects of attenuation, dispersion, and high sound-pressure levels on acoustic wave distortion in horns. Frederic G. Pla (Sverdrup Technology, Inc., Mail Stop 77-6, NASA Lewis Research Center, Cleveland, OH 44135) and Gerhard Reethof (Noise Control Laboratory, The Pennsylvania State University, 157 Hammond Building, University Park, PA 16802)

High-power sound sources have received a lot of attention in the past few years due to renewed interest in industrial applications of high-intensity sounds such as the acoustic agglomeration of aerosols or combustion enhancement. Most high-power sound sources require a horn to match the source impedance to the medium where the sound is radiated. Such horns introduce distortion in the initial waveform, which can be detrimental to the agglomeration or combustion enhancement process. Boundary-layer attenuation smooths the wave shape while dispersion breaks up the symmetry of the waveform. Horn-induced dispersion is usually the dominant dispersion mechanism, resulting in strong peaks in the waveform. Finally, due to the very high acoustic levels at the horn throat, finite-amplitude effects are responsible for a significant amount of distortion at high frequencies. Simple examples of waveform distortion due to these various mechanisms are shown. The effects of sound-pressure level, horn design, and frequency on distortion are illustrated for an exponential horn and several initial wave shapes. Experimental results are presented that compare very well with theory.

9:42

D7. Oblique reflection of a finite-amplitude dilatational wave in an elastic half-space. Kun-Tien Shu and Jerry H. Ginsberg (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

A ray theory for two-dimensional, finite-amplitude acoustic waves forming a mode within a hard-walled rectangular waveguide was described previously [K. T. Shu and J. H. Ginsberg, J. Acoust. Soc. Am. Suppl. 1 **83**, S1 (1988)]. The present paper extends those developments to the treatment of oblique reflection and mode conversion of a finite-amplitude dilatational wave (P wave) at the stress-free boundary of an elastic half-space. Due to nonlinear self-action, cumulative growth of second harmonics occur in the incident and reflected P waves in proportion to the square of the amplitude of the first-order incident and reflected P waves, respectively, but such growth is not encountered in the reflected vertically polarized shear wave (SV wave), nor in the many waves arising from nonlinear interaction between dilatational and shear waves. Uniformly valid expressions for strain are obtained by using renormalization techniques along the rays. The analysis indicates that the mode conversion between the nonlinear P and SV waves can be described by linear reflection theory. As a consequence of the reflection process, the nonlinear effect in the reflected P wave corresponds to a simple planar wave that originated from a weaker source at a longer range, even though the phase of that wave is governed by the actual propagation distance from the source. [Work supported by NSF and ONR.]

9:54

D8. Absolute measurement of particle velocity of time-harmonic acoustic waves. Joseph Vignola, Jacek Jarzynski, and Yves H. Berthelot (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

Experiments are being conducted to measure by optical means the amplitude and the phase of a steady-state, time-harmonic compressional wave being produced inside a standing wave tube filled either with air or with water. The technique used is an extension of laser Doppler anemometry in acoustics developed by Taylor [J. Acoust. Soc. Am. **59**, 691–694 (1976)]. It consists of illuminating with laser beams a small probe volume in water in which slowly drifting suspended microparticles are moving in phase with the acoustic field. The light scattered from the particles is Doppler shifted and contains information about the amplitude and the frequency of the particle motion. Preliminary results indicate that signal analysis performed on a spectrum analyzer is probably not the optimum processing technique, mainly because signals are being analyzed and displayed even at times when there is no particle in the probe volume to scatter the laser light. Instead, more meaningful and repeatable results can be obtained by using a digital data acquisition system triggered to capture the signal only when light is being scattered from a particle. The data are then transferred to a computer for further processing. [Work supported by ONR.]

10:06

D9. Application of variational principles to the evaluation of axisymmetric surface pressure and displacement along a harmonically vibrating elastic shell. Pei-Tai Chen and Jerry H. Ginsberg (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

An earlier paper [J. H. Ginsberg and P. T. Chen, J. Acoust. Soc. Am. Suppl. 1 **82**, S1 (1987)] employed assumed modes to describe the displacement and pressure along an elastic plate. The amplitudes of those modes were obtained by simultaneously satisfying Hamilton's principle for the structure and a variational principle for the pressure distribution along a vibrating body. Here, the method is extended to an elastic shell structure in the form of an arbitrary body of revolution. The energy expressions for the shell are based on Love's assumptions, while the stationary quantity for the surface pressure is derived from the Kirchhoff–Helmholtz principle. Two important facets of the derivation are the treatment of singularities in the latter and the reciprocal nature of the coupling between the pressure and surface displacement. A numerical example compares the results obtained from the variational formulation with Hayek's analytical solution [J. Acoust. Soc. Am. **40**, 342–348 (1966)] for the radiation of a spherical shell in an acoustic medium under a harmonic, concentrated force. [Work supported by the Office of Naval Research, Code 1132-F.]

10:18

D10. On left- and right-circularly polarized waves in isotropic noncentrosymmetric elastic media. Akhlesh Lakhtakia, Vasundara V. Varadan, and Vijay K. Varadan (Department of Engineering Science and Mechanics, The Pennsylvania State University, University Park, PA 16802)

Acoustic waves in solids can discriminate between a chiral scatterer and its mirror image. Thus it is possible to construct an acoustically chiral composite medium by embedding chiral microstructures in a host medium. The microstructure size should be large enough compared to the shear wavelength in the matrix medium so that an incident wave can sense its handedness; at the same time, the microstructure size should be small enough that, at least in some frequency range, the composite structure should appear to be effectively chiral. Isotropic composite media with chiral microstructure can be modeled as noncentrosymmetric (hemitropic) micropolar elastic solids, which have been the subject of some recent investigations. The simplest possible constitutive equations have been obtained, and the dispersion equations have been derived and studied. Approximate solutions of the inhomogeneous field equations have also been derived using dyadic algebra. [Work supported by the Research Center for the Engineering of Electronic and Acoustic Materials.]

10:30

D11. Detection of quantization distortion in digital audio systems. Kanako Maeda, Tsuyoshi Usagawa, and Masanao Ebata (Faculty of Engineering, Kumamoto University, 2-39-1 Kurokami, Kumamoto, 860 Japan)

In recent digital audio systems, 16-bit uniform quantization is employed in most cases, because a dynamic range of 96 dB is considered satisfactory for most cases. Ideally speaking, a dynamic range of more than 120 dB is required in music reproduction. However, harmonic distortion due to quantization can be detected when fewer bits are used for the representation of the digital signal that is reproduced at a higher sound level. In this paper, the thresholds of distortion are measured using a stationary tone as a function of sound level, the bit width of the signal, and the tone spectrum. Whether harmonic distortion is detectable when the maximum level of 16-bit digital signal is set at 120 dB SPL is examined. The effect of a dither is also examined. The results show that the dither is not always effective in reducing distortion. Furthermore, using a complex tone with exponential decay, thresholds for various conditions are measured and the results from the viewpoint of masking effects are discussed.

10:42

D12. Time-domain scattering from tungsten-carbide targets. H. Überall (Department of Physics, Catholic University, Washington, DC 20064), J. W. Dickey (David Taylor Naval Ship Research and Development Center, Annapolis, MD 21402), M. F. Werby, and Michael D. Collins (Naval Oecan Research and Development Activity, Stennis Space Center, MS 39529)

Scattering calculations are almost exclusively performed in the frequency domain. Although form functions are computed regularly, they are rarely utilized to obtain time-domain results. This is surprising because the payoff for the small amount of extra effort required for Fourier synthesis includes results that are easy to interpret in terms of causality. This approach will be used to study resonances of tungsten-carbide spheroids for aspect ratios ranging from 1–6 and for $kL/2$ up to 26. Tone bursts, cw pings, and Gaussian sources will be utilized to isolate resonances and to determine their nature. An analysis of arrival time as a function of aspect ratio gives credibility to the interpretation that resonances induced along the axis of symmetry are due to Rayleigh waves as first proposed by Flax *et al.* [J. Acoust. Soc. Am. **71**, 1077–1082 (1982)]. [Work supported by the Naval Ocean Research and Development Activity.]

10:54

D13. Numerical calculations of echo patterns in ultrasonic detection. Li Donglin, Cai Chongcheng, and Jiang Nanxiang (Five Department, Harbin Shipbuilding Engineering Institute, Harbin, People's Republic of China)

A calculation method has been developed to calculate the echo patterns of an ultrasonic pulse in the time domain. Another method for forming the acoustic radiation impedance function of the ultrasonic transducer used will also be presented. Typical numerical results are compared with experimental data, and they are in good agreement with each other.

Session E. Physical Acoustics I: Nonlinear Acoustics, Part I

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David T. Blackstock, Cochairman
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Austin, Texas 78713-8029

Invited Papers

8:30

E1. Nonlinearity in sound beams, with application to the scattering of sound by sound. Jacqueline Naze Tjøtta and Sigve Tjøtta (Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78713-8029 and Department of Mathematics, The University of Bergen, 5007 Bergen, Norway)

This paper addresses the fundamental theory of nonlinear acoustics in a thermoviscous fluid. Emphasis is given to the combined effects of nonlinearity, absorption, and diffraction in sound beams. An overview is presented of the various mathematical models used for the propagation of a high-intensity sound beam, with a discussion of their range of validity. Some examples of nonlinear effects in a sound beam are shown. Nonlinear interaction between two real sound beams is also considered. The sources may possess arbitrary phase and amplitude shading, and different sizes and orientations. The obtained results are related to earlier works on the scattering of sound by sound, which are discussed. [Work supported by the IR&D program of ARL:UT, and VISTA/STATOIL, Norway.]

8:55

E2. New measurement techniques developed on the basis of nonlinear acoustics. Takuso Sato (The Graduate School at Nagatsuta, Tokyo Institute of Technology, 4259 Nagatsuta, Midori-ku, Yokohama, 227 Japan)

When nonlinear interactions between a medium and acoustic waves are used in addition to linear interactions, two main directions can be expected in the development of new measurement techniques: (i) the extraction of new characteristics of the medium, and (ii) the construction of completely new or more compact measuring systems. In this paper, a few applications are shown of nonlinear acoustics, picked up from a method based on the use of impulsive and high-pressure pumping waves to generate the required nonlinear effects and high-frequency, small amplitude continuous probing waves to detect the results. The applications include (i) measurements of gas flow velocity and temperature distributions, (ii) nondestructive measurement of the stress distribution in metals, and (iii) imaging of dynamic characteristics, including the nonlinear parameter of soft tissues as a new means of medical diagnosis. The exact construction of each system and the corresponding experimental results are shown in detail.

Contributed Papers

9:20

E3. Theoretical approach to the virtual source based on the distortion in finite-amplitude waves. Yoshiaki Watanabe, Takao Tsuchiya, and Yasumasa Urabe (Department of Electronics, Doshisha University, Kamigyo-ku, Kyoto, 602 Japan)

An analytical approach is presented to derive the intensity of a virtual source from the viewpoint of the distortion in the finite-amplitude wave. The distortion in the arbitrary waveform over a small distance is discussed in connection with the local sound velocity which depends on the local particle velocity. From the variation in the particle velocity due to the waveform distortion, the intensity of the virtual source is derived. It is clearly shown that the virtual source can be derived using only distortion in the time domain. The present analytical approach will be useful in understanding phenomena of nonlinear acoustics.

9:32

E4. A method of measuring the nonlinearity parameter B/A of a medium using the angle dependency of β. Takao Tsuchiya, Yoshiaki Watanabe, and Yasumasa Urabe (Department of Electronics, Doshisha University, Kamigyo-ku, Kyoto, 602 Japan)

A new method of measuring B/A using a parametric receiving array is presented. The coefficient of nonlinearity β is treated exactly and is divided into two physically different terms. The effective value of β_e ($=\cos\theta + B/2A$) is defined as the function of the crossing angle θ of two waves. The angle-dependent pattern of β_e is applied to the measurements of B/A. The following two practical methods are presented: (1) using the noninteracting angle θ_n [$=\cos^{-1}(-B/2A)$] and (2) using the front–back ratio of β_e. The measurements were carried out using the two above methods for a gaseous medium and using (2) for water. The array lengths of the probing wave used were 90 mm and 40 kHz for gas, and 40 mm and 5 MHz for water, respectively. The phase modulation of the probing wave was observed and the angle dependency of β was detected. It was found that the estimated values of B/A showed good agreement with theoretical values and with the observed results presented by the other researchers.

9:44

E5. Measurement of the nonlinearity parameter B/A using the pulse-echo technique. Iwaki Akiyama (Department of Electrical Engineering, Sagami Institute of Technology, Tsujido Nishi-kaigan, Fujisawa, 251 Japan), Masato Nakajima (Department of Electrical Engineering, Keio University, Yokohama, 223 Japan), and Shin'ichi Yuta (Institute of Information Science and Electronics, University of Tsukuba, Tsukuba, 305 Japan)

A method for the measurement of the nonlinearity parameter B/A in biological tissues using the pulse-echo technique was studied. Since a finite-amplitude method is valid for the measurement of B/A in biological tissues, the values of B/A have been experimentally measured *in vitro* for several biological tissues. For medical applications it is necessary to have *in vivo* measurements, and thus it is desirable to use the pulse-echo technique. In this study, the second harmonic component of the echo signal was analyzed, and then it is shown that the rate of increase of the second-harmonic components with respect to time provides the value of B/A. This rate is, however, influenced by the attenuation coefficient and the reflection coefficient in biological tissues. In order to eliminate these effects, another pulse was used whose center frequency was twice as high as that of the former pulse. Also, the ratio of the second harmonic component of the echo signal of the former pulse and the fundamental component of the echo signal of the latter pulse is calculated. The value of B/A was determined by the rate of increase of the resulting quantity. The experiments were conducted to verify the feasibility of this technique, and the resulting values were in agreement with values from the literature.

9:56

E6. Parametric measurement of sound source directivity: Effect of the source nearfield. John D. Sample (Applied Research Laboratories, The University of Texas at Austin, P. O. Box 8029, Austin, TX 78713-8029)

The suggested use of a parametric receiving array to measure the directivity of a distributed sound source in a multipath environment is complicated by the disparity between the beam patterns of the difference frequency and linear sound fields. This disparity can be understood by considering the interaction volume as two independent sources: one representing the volume in the nearfield of the source and the other representing the farfield volume. The interaction volume in the farfield of the source has the same directivity as the linear radiation of the source, but the measured difference frequency directivity is contaminated by the contribution from the nearfield volume. The nature of the contamination in the case of a Gaussian-shaded pump interacting with an approximation of a circular piston is analyzed using a previously developed computer model. The nature of the differences between linear and difference frequency directivities is described and the dependence on range and pump location is studied. The cause of these differences is identified as difference frequency sound produced in the nearfield.

10:08

E7. Nonlinear interaction of collinear sound beams in the nearfield. Tomoo Kamakura and Yoshiro Kumamoto (University of Electro-communications, 1-5-1 Chofugaoka, Chofu, 182 Japan)

Many spectral components generated by nonlinear interaction of two collinear sound beams can be calculated using Aanonsen's method [J. Acoust. Soc. Am. **75**, 749–768 (1984)]. Aanonsen's initial pressure condition is extended to the case where a radiating wave at the source consists of two adjacent harmonics. Numerical computations are performed for a nearfield of a Gaussian source in air. Propagation curves and beam patterns of the primary and secondary waves are given for various source levels. When the source intensity is increased, the higher frequency primary wave fades out more than the lower frequency wave, and the harmonics of the difference frequency increase. Parametric generation of the difference frequency and its second harmonic component for AM excitation are also considered.

10:20

E8. A parametric array for use as an ultrasonic proximity sensor in air. Yang-Sub Lee and Mark F. Hamilton (Department of Mechanical Engineering, The University of Texas at Austin, Austin, TX 78712-1063)

An ideal ultrasonic proximity sensor has long range, high resolution, and small size. However, the absorption of sound by air makes all three characteristics difficult to achieve simultaneously with conventional acoustical designs. In this paper, the nonlinear parabolic wave equation is used to investigate the possibility of utilizing a parametric array to incorporate all three desirable characteristics. To avoid difference frequency generation at the source, one of the primary beams may be radiated from a circular (disk) source, the other from a ring source that is concentric with the disk. First, the two primary wave fields are compared, with special attention devoted to radiation from a ring source. Second, the difference frequency field is analyzed as a function of source conditions. It is concluded that, for a source with a nominal radius of 1 cm, operating in air with primary frequencies of approximately 200 kHz and on-source sound-pressure levels up to 140 dB (*re*: 20 μPa), the parametric array offers no distinct advantage over conventional ultrasonic proximity sensors. [Work supported by NSF, ONR, and the Cray Research Foundation.]

10:32

E9. A nonlinear, parametrically driven, variable-reluctance electroacoustic transducer. W. B. Wright and G. W. Swift (Condensed Matter and Thermal Physics Group, Los Alamos National Laboratory, Los Alamos, NM 87545)

There is a well-known class of nonlinear acoustic drivers in which an acoustic force (or pressure) is exerted by the magnetic interaction of an electric current with itself, so that the force is proportional to the square of the current. The St. Clair driver [H. W. St. Clair, Rev. Sci. Instrum. **12**, 250 (1941)] is the best-known device of this class. Less well known is the inverse effect, the nonlinear generation of an electric current by an acoustic vibration, achieved in the same apparatus. In the present case, a resonant series LC circuit is excited by varying L (geometrically, by acoustic vibration) at twice the resonance frequency of the LC circuit. The resonance requirement and inherent overwhelming nonlinearities make this system useless as a high-fidelity microphone. But several interesting nonlinear dynamical phenomena can be observed with this simple system; and, in addition, it is capable of highly efficient power transduction. [Work supported by DOE/BES.]

10:44

E10. Propagation of the difference frequency wave generated by a truncated parametric array through a water–sediment interface. Liu Wensen and Xu Zhenxia (Institute of Acoustics, Academica Sinica, Beijing, People's Republic of China)

T. G. Muir *et al.* observed experimentally that the maximum of the received sound pressure in the sediment insonified by a parametric array departed significantly from the prediction of Snell's law. The wave fronts penetrated more steeply into the sediment and the attenuation with depth was less than predicted by plane-wave theory at the postcritical incidence. It was found that, due to the variation of the length of the parametric array and the variation of the attenuation in the sediment, the maximum of the received sound pressure occurs at the line-of-sight between the projector and the hydrophone. It has been proved both theoretically and experimentally that Snell's law is still valid when the length of the parametric array and the attenuation in the sediment are kept constant. Due to the contribution of the secondary sources close to the boundary, the postcriti-

cal penetration of the parametric array can be deeper. The lateral wave should be considered at the postcritical incidence.

10:56

E11. Nonlinear interaction of an acoustic wave and mean flow at a stagnation point (model verification). Charles Thompson and Martin Manley (Department of Electrical Engineering, Laboratory for Advanced Computation, University of Lowell, 1 University Avenue, Lowell, MA 01854)

A model has been developed to describe the nonlinear interaction of an acoustic disturbance and the mean flow over a bluff body near the stagnation point. The model identifies the neutral stability condition and predicts the evolution in space and time of disturbances in unstable flow. The available experimental data regarding this interaction are extremely limited. Two sets of data that do shed light are those of Hassler (1971) and Colak-Antic (1971). These experiments concerned the effects of unsteady disturbances on a mean flow for a geometry similar to the model geometry. The flow visualizations and hot-wire anemometer measurements of these reports illustrate some of the complex phenomena involved in the interaction. Interaction model predictions will be compared with experimental results. [Work supported by Analog Devices Professorship.]

11:08

E12. An exact solution for finite-amplitude plane sound waves in a dissipative fluid. Hideto Mitome (Mechanical Engineering Laboratory, AIST, MITI, 1-2 Namiki, Tsukuba, 305 Japan)

The propagation of finite-amplitude plane sound waves in a dissipative fluid can be described by Burger's equation, and its exact solution obtained [D. T. Blackstock, J. Acoust. Soc. Am. **36**, 534–542 (1964)]. In this paper, an exact solution for sound pressure, which is suitable for the direct numerical computation of the waveform in the time domain, is derived. Numerical results show that this solution describes the whole propagation process, including shock formation and its decay. Computation is possible up to $\Gamma = 175$, where Γ is the Goldberg number indicating the importance of nonlinearity relative to dissipation, for any value of X (distance normalized by shock formation distance). Although it becomes difficult above this value (except at larger X's) because of its functional form, the solution connects smoothly with the Fubini solution at $X<1$ and the Fay solution at $X>3.5$. Since this solution is exact and gives the waveform at any position, it can serve as a standard solution for various approximate solutions. As an application, the saturation for the fundamental component and that for the entire wave are shown.

11:20

E13. Nonlinear variation in waveform and attenuation of tone-burst sound waves with finite amplitude. Akira Nakamura (Institute of Scientific and Industrial Research, Osaka University, 8-1 Mihogaoka, Ibaraki, 567 Japan)

Propagation in a circular pipe of tone-burst waves with finite amplitude was simulated by a computer. The results are compared with some experimental results [Y. Watanabe and Y. Urabe, Proc. 11th ICA, Paris, Vol. 1, pp. 337–340 (1983)]. The attenuation is separated into two parts in the time profile of the tone-burst waveform for comparison. One is the attenuation of energy effectively corresponding to a period given by a summation of the head and tail half-periods in the time profile. The other is the attenuation per period of the sinusoidal part located between the head and tail parts. It is found that (1) the attenuations of the head plus tail are considerably smaller than the attenuation of the sinusoidal part, and agree with the theoretical values for N waves and also with the experimental results, and that (2) the attenuations of the sinusoidal part agree with the theoretical values for repeated sawtooth waves but does not agree with experimental values, because the envelope of the profile used in the experiments had been deformed into an irregularly shaped rectangle.

11:32

E14. Finite-amplitude propagation in lossless and absorptive media. Clarence R. Reilly (Department of Electrical Engineering, University of Rochester, Rochester, NY 14627 and Medical Instrumentation Program, Indianapolis Center for Advanced Research, Inc., Indianapolis, IN 46204) and Kevin J. Parker (Department of Electrical Engineering and Rochester Center for Biomedical Ultrasound, University of Rochester, Rochester, NY 14627)

The propagation of finite-amplitude waves in distilled water and an absorptive fluid media with acoustic characteristics similar to tissue was investigated. Axial and focal beampatterns of linear and quasilinear waves were obtained from lens focused sources of 1.75, 2.25, 2.94, and 3.38 MHz using a PVDF needle-type hydrophone. Experimental beampatterns were compared to spherically converging and focused Gaussian beam theories. The shock parameter at the focus predicted from Gaussian theory was in general agreement with the estimate made from the relative strengths of the first four harmonics to the fundamental. Apodization provided by the lens reduced, but did not eliminate, the nearfield maxima, minima, and sidelobes associated with a piston source. Additional sidelobes predicted from the Khokhlov–Zabolotskaya–Kuznetsov (KZK) equation were evident in the harmonic beampatterns. Harmonic focal beamwidths decreased as $n^{-1/2}$ for the lower source frequencies, in agreement with theory. Beamwidths were generally more narrow in the absorptive media than in the water. The second harmonic peak location on-axis was distal to the fundamental, although the difference was small for higher source frequencies and propagation in the absorptive media. Higher harmonics had sharper axial focal dimensions, but the position of the axial peak was dependent on the trade-off between propagation growth and decay caused by attenuation. The extension of the results to propagation in tissue is noted.

Session F. Physiological Acoustics I: Hair Cell Transduction and Cochlear Frequency Analysis

Jozef J. Zwislocki, Cochairman
Institute for Sensory Research
Syracuse University
Syracuse, New York 13244

Seiichiro Namba, Cochairman
Department of Psychology
Osaka University
Toyonaka, 560 Japan

Introduction

8:30

F1. Introduction: Hair cells as integral parts of cochlear mechanics. Jozef J. Zwislocki (Institute for Sensory Research, Syracuse University, Syracuse, NY 13244-5290)

The last 20 years brought about a revolution in our concepts of cochlear sound processing. Sound selectivity proved to be much greater in live animals than had been found by Békésy in *post-mortem* preparations, and the discovery of an active biological process in the cochlea has provided a partial explanation for the difference. The active process seems to have been accounted for by the discovery of an electrically and biochemically controlled motility of the outer hair cells. In addition, the demonstrations that the stereocilia of cochlear hair cells are stiff and the tectorial membrane is compliant by comparison, together with other insights, suggest that the classical model of hair-cell stimulation must be radically modified. A new model has been proposed, which is consistent with the current experimental evidence. The introduction, the two distinguished lectures, and the following invited papers review some key aspects of the still ongoing revolution. They focus on the electromechanical processes in the hair cells, as examined *in vitro* and in the cochlear environment.

Distinguished Lectures

8:55

F2. Mechanoelectrical transduction by vertebrate hair cells. A. J. Hudspeth (Department of Physiology, University of California School of Medicine, San Francisco, CA 94143-0444)

The initial cellular stage in the ear's response to sound is the transduction of mechanical energy into electrical signals of hair cells of the internal ear. This presentation will summarize *in vitro* experiments on the biophysical basis of transduction and will present a model for the process. Mechanoelectrical transduction rests upon the activity of transduction channels, relatively nonspecific, cation-permeant ion channels. These mechanically sensitive channels, which number about 100 per cell, appear to lie near the distal tip of the hair bundle. Excitatory stimuli cause transduction channels to open with a latency of only a few microseconds; transduction is therefore unlikely to require a second messenger. The dependence of the channels' rate of response upon the stimulus amplitude suggests that stimulation affects the rate constants for channel opening and closing. Mechanical measurements of changes in the bundle's stiffness during channel gating support a model for transduction in which excitatory stimulation acts through an elastic linkage in the hair bundle to open each channel's gate.

9:40

F3. The frequency limits to transduction in hair cells of the turtle cochlea. Robert Fettiplace (Physiological Laboratory, University of Cambridge, Cambridge, United Kingdom)

During transduction in auditory hair cells, displacements of the hair bundle modulate an inward transducer current; this then evokes a receptor potential that is filtered by the electrical properties of the cell membrane. The frequency characteristics of these steps have been studied by recording membrane currents in isolated turtle hair cells. When the hair bundle is rapidly deflected, the transducer current is activated with a maximum time constant of about 0.1 to 0.2 ms at 23 °C, thus faithfully encoding frequencies below 1 kHz. The ensuing receptor potentials are shaped by a sharply tuned resonance resulting from the interaction of a voltage-dependent Ca^{2+} current and a Ca^{2+}-activated K^+ current. The activation time constant for the K^+ current varies inversely with, and largely determines, the resonant frequency of the hair cell in the range 20 to 600 Hz. The time constant of the Ca^{2+} current (0.3 ms) is rapid compared to the K^+ current but must impose a corner frequency of about 500 Hz on transmission across the afferent synapse.

10:25–10:30
Break

Invited Papers

10:30

F4. Mechanoelectrical transduction of the chick hair cell. Harunori Ohmori (National Institute for Physiological Sciences, Okazaki, 444 Japan)

Hair cells transduce mechanical energy applied to the apical hair bundle into an electrical signal, i.e., the transduction potential, through gating of mechanically gated ion channels (m-e.t. channel). The gate demonstrates sigmoidal displacement versus response relationship irrespective of the variation in the hair bundle's length (from 7 to 30 μm), as long as the stimulus is applied to the hair bundle at a fixed height from its insertion into the apical surface of the cell. The angular displacement of the hair bundle could therefore be a primary factor that determines the gating of the m-e.t. channel. The channel demonstrates strong selectivity to divalent cations, particularly to Ca ions, over monovalent cations. Mn is one of the permeants through the channel. Fura-2 fluorescence is modified by Ca and Mn ions permeated through m-e.t. channel, and these ions produce a localized peak and a localized depression, respectively, in the 340/380-nm fluorescence ratio images during mechanical stimulation in the vicinity of the hair bundle's insertion into the apical surface of the cell. This might be indicating the site of the mechanoelectrical transduction.

11:00

F5. *In vivo* and *in vitro* studies of intracochlear electrical potential gradients. W. E. Brownell (Departments of Otolaryngology–NNS, and Neuroscience, The Johns Hopkins University, Baltimore, MD 21205)

Currents of similar magnitude to those measured *in vivo* evoke rapid length changes of nearly 1 μm in isolated outer hair cells (OHCs) maintained *in vitro*. The ability of OHCs to follow electrical stimulation at greater than 8 kHz makes OHC electromotility a strong candidate for the active mechanical process assumed to contribute to the exquisite tuning and sensitivity of the intact cochlear partition. An attempt is made to identify the biophysical mechanism responsible for these rapid movements by measuring the length changes evoked by potential gradients produced with two tight seal whole cell electrodes. During these experiments, a second, considerably slower, type of electrically induced OHC shape change has been observed. The mechanism responsible for this slow, reversible, electrically induced volume change appears unlike that responsible for the fast electromotility. [Work supported by NIH (NS23567) and ONR (441k704).]

11:30

F6. Nonlinearities seen in hair-cell data from *in vivo* and *in vitro* preparations. Peter Dallos, Burt N. Evans, and Mary Ann Cheatham (Auditory Physiology Laboratory, Northwestern University, Evanston, IL 60201)

The growth patterns of fundamental, dc, and harmonic receptor potential components recorded *in vivo* from the low-frequency region of the guinea pig cochlea as a function of stimulus frequency versus best frequency of the cell were examined. Changes in these components with natural variations in membrane voltage and with electrical polarization of the cell are also considered. A report is also given on nonlinear growth patterns of outer hair-cell motile responses as a function of cell length (best frequency).

12:00–12:10

Discussion

Session G. Speech Communication I: Analysis and Synthesis Part A (Poster Session)

Astrid Schmidt-Nielsen, Cochairman
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Fumitada Itakura, Cochairman
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Contributed Papers

Posters must be set up before 8:00 a.m. (before Opening Plenary Session). All posters will be displayed from 8:30 to 10:00 a.m. To allow contributors the opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:30 to 9:15 a.m. and contributors of even-numbered papers will be at their posters from 9:15 to 10:00 a.m.

G1. Efficient subband coding of speech with optimized uniform DFT filter banks. A. Satt and D. Malah (Department of Electrical Engineering, Technion, Haifa 32000, Israel)

Uniform DFT filter banks (FB) offer reduced implementation complexity of subband coders (SBC) as compared with QMF-based SBC. However, because known design techniques aim at minimizing the overall response error of the FB, using either deterministic [V. K. Jain and R. E. Crochiere, Proc. ICASSP **83**, 228–231 (1983)] or statistical [A. Dembo and D. Malah, IEEE Trans. Acoust. Speech Signal Process. **ASSP-36**, 328–341 (1988)] error measures, the performance of DFT-based SBC was found to be subjectively inferior. A new approach is presented for DFT-FB design, which is based on a frequency-doman distortion function in which different weights are applied to the different error components (including interband aliasing and leakage of both signal and quantization noise) thus allowing an optimized design for SBC. The minimization of the proposed distortion function results in two sets of linear equations that are solved iteratively to obtain the optimal analysis and synthesis prototype filters. A 16-band DFT-based SBC operating at 16 kb/s with the designed filters was found in simulations to have subjective and objective performance similar to a QMF-based SBC at the same rate, but with less than half of the computations.

G2. Multimode coding: A novel approach to narrow- and medium-band coding. Tomohiko Taniguchi, Shigeyuki Unagami (Speech Signal Processing Section, Fujitsu Laboratories Ltd., 1015 Kamikodanaka, Nakahara-ku, Kawasaki, 211 Japan), and Robert M. Gray (Information Systems Laboratory, Stanford University, Stanford, CA 94305)

Most research on narrow-band coding is concentrated on how to transmit excitation parameters efficiently. However, the important thing in obtaining a good reproduced speech quality is how to control the balance of the transmission bit rate between the excitation and the LPC parameters. Multimode coding, which is proposed here, has two coding modes: One is a mode that transmits the LPC parameters in every frame, as conventional coders (A mode). The other is a mode that avoids the transmission of the LPC parameters by using the same coefficients as the previous frame and increases the bits allocated to the residual quantization instead of to the LPC transmission (B mode). In each frame, the mode selection takes place based on an evaluation of the reproduced speech quality, and the assignment of transmission information is dynamically controlled by switching between the two modes. This coding algorithm is applied to a 7.2 kb/s CELP coder, and approximately 3 dB of improvement is achieved in SNR compared with a conventional CELP coder. The B mode was used 78%–82% of the time.

G3. Realization of a multirate speech codec utilizing spectrum peak emphasis. Tetsu Taguchi (Radio Application Division, NEC Corporation, Fuchu, 183 Japan)

One of the most promising means of realizing multirate speech coding is multipulse excited linear predictive coding (MPELPC) adopting an auditory weighting filter (AWF). This AWF effectively shapes quantization noise to be masked by the speech signal and ameliorates speech quality at approximately 8 kb/s or more. However, below this rate, the AWF is not effective because the SNR is so low that the signals scarcely mask the noise. When the SNR is rather low, only the high-power frequency components of speech signals (whose levels are higher than those of the noise) should be encoded. In order to emphasize these components effectively upon encoding, a spectrum peak emphasis (SPE), which is performed by a prefilter and utilized in MPELPC, has been proposed. Below 8 kb/s, it has been shown by computer simulation that compared to the AWF, the SPE improves SNR at least by 2.2 dB. Subsequently, a multirate code from 16 to 4.8 kb/s, basically utilizing the AWF but with the SPE below 8 kb/s, has been realized on a DSP, NECμPD77230. This coder is applicable to mobile satellite communication.

G4. Speech coding by Model Reference Adaptive Control. Kiyoshi Hashimoto and Makoto Yasuhara (The University of Electro-Communications, Chofu, 182 Japan)

Model Reference Adaptive Control (MRAC) coding [M. Yasuhara, Trans. IEICE **E71**(1), 34–42 (1987)] was applied to speech signals. The primary feature that characterizes MRAC coding is that it is not a predictive coding, but control coding. It determines the control signal so the output of a plant (an ARMA speech production model) follows the speech waveform. The control signal is quantized and transmitted to the receiver. Since at the receiver only a noisy state vector of the plant is observable, a Kalman filter is embedded in the plant to estimate the optimal state vector. Experiments were conducted to evaluate the transmission performances in the S/D ratio for the MRAC coding in comparison with those for the forward ADPCM. With a rate of 2.25 bits/sample, the AR order of the plant between 5 and 12, and the MA order of 1, the average performance for the MRAC coding is around 20 dB, which is 6 dB higher than for the FADPCM.

G5. On improving voice periodicity prediction in codebook-excited LPC coders. Daniel Lin (International Mobile Machines, 2130 Arch Street, Philadelphia, PA 19103)

Codebook-excited predictive coders are able to synthesize high-quality speech at bit rates of 8 kbits/s and above. However, at lower bit rates (e.g. 4.8 kbits/s), speech enhancement techniques, such as adaptive postfiltering, are needed to improve subjective performance [cf. P. Kroon and B. S. Atal, Proc. ICASSP (1987)]. The main source of quality degradation at 4800 b/s is due to the reduced transmission bandwidth of the excitation parameters, which results in an inaccurate representation of the LPC innovation signal. In this paper, methods are proposed for improving the voiced speech excitation and prediction in the codebook-excited coder. The modification uses a composite (i.e., product) excitation codebook together with a close-loop pitch prediction filter that is optimized relative to the source codebook. Several optimization creiteria for the source and predictor are compared in terms of performance and computational complexity. The best result shows a segmental SNR improvement of nearly 2 dB over the conventional stochastically excited LPC coder.

G6. Cascaded likelihood vector compression of linear predictive models. Yair Shoham (AT&T Bell Laboratories, 600 Mountain Avenue, Murray Hill, NJ 07974)

Compression of linear predictive spectra has applications in acoustics research, particularly, in communication of acoustical information. Cascaded likelihood vector compression (CLVC) is proposed for coding the spectral parameters of linear predictive models at rates of 20 to 26 bits per model. The system is based on representing the LP all-pole model as a cascade of two lower-order models. The partitioning of the LP polynomial is done in the root domain by clustering the roots into two distinct groups. The compressor uses two codebooks to quantize each of the lower-order models. However, the quantization process is done so as to jointly optimize the overall performance. The likelihood ratio distortion measure is used as a performance criterion. Splitting the LP model into two subsystems dramatically reduces the complexity while the efficiency of vector compression is essentially preserved. Experimental results show an average performance of 1.59, 1.41, 1.24, and 1.10 dB of log-spectral distortion at the rates of 20, 22, 24, and 26 bits per model, respectively. These results indicate high-quality model compression particularly for communication applications where the number of bits assigned to the LP model is very limited.

G7. Temporally overlapping segment quantization for speech coding. Masaaki Honda (Information Science Laboratory, NTT Basic Research Laboratories, 3-9-11 Midori-cho, Musashino, 180 Japan)

A method for the efficient segment quantization of LPC spectral parameters is presented for very low-bit-rate speech coding. Recent methods of segment quantization for speech coding have used a concatenation segment model, in which spectral parameters are represented as a concatenation of variable-length code spectral segments. The present method uses a temporally overlapping segment model to capture more efficiently the variance in the spectral parameters due to coarticulation effect in speech. Spectral parameters are represented as a sequence of temporally overlapping segments, in which each code segment is decomposed into fixed-length temporal patterns and their weighted patterns. A dynamic programming procedure is used to determine the optimum code segments and the segment positions of the overlapping model so as to minimize the spectral distance between the input and the model spectral parameters in the analysis window. An iterative algorithm for code segment design in this model is also presented. Comparison of the overlapping model with the concatenation model for speech coding [Shiraki and Honda, to appear in IEEE Trans. ASSP] is described in terms of spectral distortion and computational complexity.

G8. Noise suppression effect of a linear predictor and a coding scheme for narrow-band speech communications. Yasuhiko Arai and Toshio Yagi (Corporate Engineering Division, Matsushita Communication Industrial Company, Ltd., 600 Saedo-cho, Midori-ku, Yokohama, 226 Japan)

The relation between the perceptual noisiness of quantization error and the number of prediction coefficients was experimentally studied. It was revealed that a speech coder (a kind of APC) with ten coefficients shows considerably less noisiness than that one with four coefficients. Though improvement of the segmental SNR was only 1 dB, perceptual noisiness was improved by an equivalent value to a quantizing accuracy of 1.5 bits (approximately 9 dB). Taking this result into consideration, a coder scheme was proposed for special-purpose, narrow-band, wireless communcations. The narrow-band speech coder to be used in very noisy channels having a transmission error rate of 10^{-2} or 10^{-1} requires good intelligibility and less perceptual noisiness. It was confirmed that a source coder having eight or ten prediction coefficients and an error correction code for side information would meet the requirements. Even if the residual information fails to be transmitted, the source decoder is capable of synthesizing intelligible speech from only the side information including coefficients.

G9. Effects of degradation in vowel steady-state and transition on the perception of coded speech. Kazunori Ozawa and Eisuke Hanada (C&C Information Technology Research Laboratories, NEC Corporation, 4-1-1 Miyazaki, Miyamae-ku, Kawasaki, 213 Japan)

In order to investigate the perceptual importance of the vowel steady state and transition on the perception of coded speech, and to obtain cues for further reducing the bit rate in speech coding methods, coded speech was systematically deteriorated during vowel steady states and transitions. Multipulse excited speech coding (MPC) [T. Araseki and K. Ozawa, Proc. GLOBECOM, 794–798 (1983)] at 16 kb/s was used as a speech coding method. Perceptual experiments were carried out using 18 kinds of meaningless sequences having a Japanese CVCV structure including 5 vowels, 2 fricatives, and 6 stops. The intelligibility of the vowels and consonants was measured. The results showed that the intelligibility for vowels was the same as that of the original 1 kb/s MPC, even when all of the multipulse excitation information during the vowel steady states was deleted and the multipulse information during transition parts was reduced to $\frac{1}{4}$ of the 16 kb/s MPC. The intelligibility began to decrease when the multipulse information during the transitions was reduced to $\frac{1}{8}$ of the 16 kb/s MPC. The results, including the intelligibility of the consonants, will be discussed in detail.

G10. Speech quality assessment in low-bit-rate speech coding taking into consideration quality variation among speakers. Hiromi Nagabuchi (NTT Telecommunication Networks Laboratories, 3-9-11 Midori-cho, Musashino, 180 Japan)

A method for assessing the quality of low-bit-rate speech coding is proposed. This method considers the speaker dependency of coded speech quality, which has not been done before. First, it is shown that coded speech quality varies with speakers more in low-bit-rate coding than in PCM. Next, the power of the linear prediction error of the signal of speech weighted by speech power, G_{opt}, is introduced as a measure for estimating quality variation among speakers. It is shown that G_{opt} can be used as an effective index for selecting speakers in coded speech quality evaluation. Finally, a speech quality assessment method is proposed in which (1) test speakers are selected to cover uniformly the whole range of the G_{opt} distribution; and (2) the assessment results are expressed in terms of not only the average but also the standard deviation of measured values for each speaker weighted according to the G_{opt} distribution characteristics. This method can assess the performance of low-bit-rate speech coding more accurately due to the increased reproducibility of assesed values when applied to different speakers.

G11. Spectral estimation of speech based on a composite cosine wave model. Shuzo Saito and Kazuhiro Tamaribuchi (Department of Electronics, Kogakuin University, 2665-1 Nakanocho, Hachioji, Tokyo, 192 Japan)

An algorithm of an analysis for acoustic signals composed of cosine waves and its application for speech signals are reported. It is shown theoretically that when the number of the frequency components of an acoustic signals is m, only $3m$ discrete sequential data are needed for frequency analysis. It was found that a kind of analysis-by-synthesis technique on the signal waveforms is needed for the analysis of the amplitude-quantized data of the speech signals, setting the allowable error limit on the reconstructed signal amplitude, which is usually set at 0.5 LSB in this paper. This method of analysis is applied to the frequency analysis of several artificial acoustic signals as well as natural speech signals. These signals are sampled at 10 kHz and their amplitudes are quantized at 12 bits and then analyzed. Results show that this method can be used for amplitude quantized acoustic/speech signals under a procedure where the number of frequency components m is set first at a rather large value, e.g., 15, and then an analysis-by-synthesis technique is applied recursively to the reconstructed waveform using the reduced number of frequency components and/or an appropriate allowable error limit.

G12. Application of the Wigner distribution to speech analysis. Andrew Wilson Howitt (Room 36-511, Research Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge, MA 02139)

The Wigner distribution, a time-frequency signal representation similar to the spectrogram, has recently been applied to speech analysis. It offers higher resolution than the spectrogram, but introduces artifacts that do not correspond to the components of the signal. These artifacts seem to be characterized by negative values of the Wigner distribution. If so, skilled experimental subjects should be able to visually identify and disregard the artifacts. Subjects examined spectrograms and Wigner distributions of a set of synthetic speech utterances, and performed a simple formant-tracking task (initial slope estimation) on them. The subjects performed this task marginally better using the Wigner distribution than the spectrogram. This performance advantage of the Wigner distribution held over the range of formant trajectories that the synthesizer could reliably produce. Performance depended critically on the subjects' understanding of artifacts. A theoretical prediction of a limit on formant slope, previously untested, was consistent with both spectrograms and Wigner distributions of natural speech. The overall conclusion is that the Wigner distribution is a viable alternative to the spectrogram for analysis of rapid spectrum changes. However, the burden of its greater complexity seems to outweigh its potential performance advantages.

G13. A comparison of alternative procedures for digital spectral analysis of speech. Alex Kania, Donald G. Jamieson, and Ketan Ramji (Department of Communicative Disorders, University of Western Ontario, London, Ontario N6G 1N4, Canada), and Terrance M. Nearey (Department of Linguistics, University of Alberta, Edmonton, Alberta T6G 2H1, Canada)

Researchers now enjoy an increasingly wide selection of alternative procedures with which to characterize the time $\times$ frequency $\times$ amplitude variation in a speech waveform. Spectral analyses using a filter-bank technique, the digital Fourier transform (DFT), and a variety of linear predictive coding (LPC) procedures are now widely available. Approaches using an autoregressive (AR), moving average (MA), autoregressive moving average (ARMA), or maximum entropy (ME) procedure are already widely available, or are becoming available. The present paper reviews the assumptions of, and the interrelations between, alternative methods, and outlines the relative advantages of the various approaches in providing information about the speech waveform. Finally, the human factors aspects of alternative spectral displays are discussed.

G14. A recursive estimation of ARMA parameters based on a robust time-varying model for speech analysis. Masahiro Serizawa, Nobuhiro Miki, and Nobuo Nagai (Research Institute of Applied Electricity, Hokkaido University, N12-W6 Sapporo, 060 Japan)

This paper presents a recursive estimation of ARMA parameters based on a robust time-varying model for speech analysis. This algorithm is basically similar to the recursive least-squares estimation (RLS), but it is different in that the time variation of the ARMA parameters is dependent on past ones. This is based on the assumption that the speech production process does not vary instantly. This method has two linear estimators: an input estimator and a parameter estimator for known input. The variation of the parameters is estimated by using the likelihood function. The proposed method is equivalent, under certain conditions, to the RLS with the forgetting factor. However, using the proposed method, this factor can be estimated as the value that represents the variation of the parameters. Finally, the proposed method was applied to a synthetic speech and real speech. The results show that the estimated spectra sufficiently represent the dynamic movement of formants without jitters or extreme estimation errors.

G15. Analysis of Kohonen's self-organizing feature map for vector quantization of speech. K.-P. Li and J. Naylor (ITT Defense Communication Division, San Diego, CA 92131)

This work presents an analysis of a vector quantization (VQ) process based on Kohonen's self-organizing feature map algorithm [T. Kohonen, *Self-Organization and Associative Memory* (Springer, New York, 1988), 2nd ed.]. This process is an unsupervised learning technique that converges to a set of ordered nodes that represent cluster centers. These ordered nodes, the VQ codebook elements, are closely related to the sample density function of the training data. The VQ codebooks generated with this algorithm are compared to those generated with a previously developed speech VQ algorithm based on K-means clustering. Performance criteria include the average distortion between input vectors and their coded values and the maximization of entropy, a measure of the distribution of codebook element usage. The Kohonen algorithm provides superior codebooks without requiring more processing. Also, because the codebooks generated by this algorithm are ordered and reflect the statistical characteristics of the training data, the codebooks are useful tools for speech analysis, as well as for speech categorization.

G16. Characteristic behavior of long-term speech vector sums with application to speaker identification. Timothy R. Thomas (Los Alamos National Laboratory, Los Alamos, NM 87545) and Wojciech Zakrzewski (Department of Mathematical Sciences, University of Durham, Durham DH1 1LE, England)

It is known that the parameters derived from linear predictive coding of speech can provide a basis for speaker identification, and that long-term averages improve the reliability of such recognition schemes. The present study obtained long-term descriptors of speech by converting 20-ms frames into unit direction vectors in 14-dimensional cepstral coefficient space. Various numbers of these vectors were then summed, either sequentially or randomly, and the statistical characteristics of the lengths and directions were explored, using both analytical and computational tools. Interspeaker and intersession comparisons were then made, as well as comparisons to sums obtained from randomly generated vectors and from frames selected on the basis of total energy. Several interesting findings were revealed, including the observation that while the average angle between frames and the frame-to-frame changes in direction remained remarkably consistent across speakers and sessions, the direction of the long-term sums was reliably different between speakers, but not sessions. Furthermore, a proper selection on the basis of total energy in the frame can enhance interspeaker differences.

G17. Evaluation of cepstral lifters for articulatory codebook search. Peter Meyer (AT&T Bell Laboratories, Murray Hill, NJ 07974)

Of great importance for the success of the articulatory approach to speech coding [Schroeter *et al.*, IEEE-ICASSP, 308–311 (1987)] is the use of a good distance measure between a given speech signal and the entries in a stored codebook of impulse responses and corresponding vocal tract shapes (articulatory codebook). One promising distance measure is the weighted cepstral distance. Since the impulse responses in the articulatory codebook do not include glottal characteristics, optimal weighting functions (cepstral lifters) are derived to reduce the influence of a varying glottal source on the cepstral distance measure. This is done by examining the ensemble of cepstral coefficients of speech produced by an articulatory speech synthesizer that also includes a vocal-cord model. The obtained cepstral lifters are optimal for the given ensemble of cepstral coefficients and for given constraints on the weighting function. They are different for cepstral coefficients derived from the power spectrum (FFT cepstra) and those derived from LPC coefficients (LPC cepstra). The performances of the obtained cepstral lifters are compared in an articulatory codebook search.

G18. A perception-based LSP distance measure for speech recognition. K. K. Paliwal (Tata Institute of Fundamental Research, Homi Bhabha Road, Bombay 400005, India)

Because of their high correlation and reduced sensitivity to quantization errors, the line spectrum pair (LSP) frequency parameters have been used recently for efficient quantization of LPC information for speech coding. In the present paper, the LSP representation is used for speech recognition and a new perception-based LSP distance measure is proposed. This distance measure exploits the following two properties of the speech perception process [D. H. Klatt, Proc. ICASSP, 1278–1281 (1982)]: (1) The formant frequencies are the most important parameters for speech perception: and (2) the formant bandwidths and spectral tilt contribute very little to speech perception. The present distance measure uses these two properties in the form of weights in a weighted Euclidean distance measure. In order to derive these weights, the LPC power spectrum $P(f)$ is computed for each speech frame and the weight for a given LSP frequency f_i is taken to be proportional to $[P(f_i)]^c$. The perception-based LSP distance measure is studied here on a speaker-dependent speech recognition task. The optimum value of the exponent c is found here to be 0.15. The perception-based LSP distance measure results in a recognition score of 95.7%, while the recognition accuracy is found to be 89.9% by using the conventional Euclidean distance measure on LSP parameters and 94.1% by using the liftered cepstral distance measure [B. H. Juang *et al.*, IEEE Trans. Acoust. Speech Signal Process. ASSP-35, 947–954 (1987)].

G19. Unsupervised speaker adaptation of spectra based on a minimum fuzzy vector quantization error criterion. Hiroshi Matsumoto and Yasuki Yamashita (Department of Electronic Engineering, Faculty of Engineering, Shinshu University, 500 Wakasato, Nagano, 380 Japan)

Reference spectral code vectors are adapted to an input speaker by adding a distance-dependent linear combination (an interpolated vector) of speaker difference vectors (unknown adaptation vectors) at given typical points in the reference spectral space. The unknown adaptation vectors are estimated so as to minimize the total fuzzy vector quantization errors for training vectors using an adapted reference codebook. This minimization is iteratively resolved under a restriction on the sum of the norms over all the unknown adaptation vectors. The above adaptation process is applied in a stepwise manner by increasing the number of typical spectral points. The fuzziness, interpolation parameters and the limits of the norm were examined through 28 word recognition tests for 4 male and 4 female speakers, using reference patterns from a male speaker. Under the best conditions, with 16 adaptation vectors, this method achieved the same recognition accuracy as a supervised speaker adaptation for male speakers with training samples as short as 3 s.

G20. An approach to noise cancellation for speech recognition systems. Shuji Kurokawa, Michiharu Mito, Yoshihiko Horio, and Shogo Nakamura (Nakamura Laboratory, Department of Electronic Engineering, Tokyo Denki University, 2-2 Kanda Nishiki-cho, Chiyoda-ku, Tokyo, 101 Japan)

This paper describes an approach to simple noise cancellation for speech recognition, such as template matching on time-spectrum patterns. This system has two inputs. The primary input I_p receives a speech signal $S(t)$ corrupted by an environmental noise $N(t)$. The second input I_r, used as a reference signal, receives $kN(t+t_d)$, that is, a correlated version of the noise in I_p, where k represents the level ratio of $kN(t+t_d)$ to $N(t)$ included in the primary signal, and t_d is the time difference between I_p and I_r. The two input signals are analyzed using a bandpass filter bank (BPFB). The noise cancellation is performed in the following manner: (1) The level ratio k is estimated; (2) the time difference t_d is also estimated using k; (3) the cancellation is performed on every output of the BPFB using k and t_d. At the same time, a speech section is detected by using the variations in each spectrum difference between the two input signals. The values of the parameters k and t_d are fixed at the beginning of the speech section while a speech signal exists.

G21. Precise detection of isolated words in noise by the linear prediction method. Hidefumi Kobatake and Katsuhisa Tawa (Department of Electronic Engineering, Tokyo University of Agriculture and Technology, 2-24-16 Nakamachi, Koganei, 184 Japan)

This paper presents a new method of detection for speech in stationary noise. First, a fundamental principle of segmentation is derived. It is shown that the final prediction error (FPE) decreases monotonically as the length of the processed signal increases, if the signal is stationary or homogeneous, and the FPE tends to increase when the homogeneity of the signal is lost. This means that the instant when the FPE is at its minimum corresponds to the position of the initial word. Based on this principle, the final prediction error is proposed as a criterion to find the speech segments in noise precisely. Second, experiments to test the performance of the proposed method are presented. Experimental results show that the proposed method is sensitive enough to detect changes in signal characteristics. A detailed comparison of the accuracy of the proposed method and that of conventional methods has been made and the effectiveness of the proposed method is shown.

G22. Evaluating the intelligibility of different speech degradations using the ICAO spelling alphabet. A. Schmidt-Nielsen (Naval Research Laboratory, Code 5532, Washington, DC 20375-5000)

A series of experiments was conducted using spelling alphabet words (ALFA, BRAVO, CHARLIE, etc.) to evaluate the intelligibility of several severely degraded voice communication conditions. Under severe speech degradation, standard consonant intelligibility tests such as the DRT and MRT are often not very helpful in predicting the actual usability of a particular voice system because consonant information is often so degraded that listeners must rely on other cues such as vowel identity, prosody, and context in order to understand the speech. The spelling alphabet is a small highly distinctive vocabulary that was designed to be robust in noise. Scores are compared with rhyme test (DRT) scores and with performance on a sentence verification task. The most frequently confused work pairs are compared across different types of degradation. For LPC with bit errors, confusions tend to be highly asymmetric (PAPA is frequently heard as ALFA, but the reverse confusion almost never occurs), whereas with noise degradation confusions are more likely to occur in both directions (e.g., ALFA–DELTA and DELTA–ALFA). Speech degraded with interrupted noise shows yet a different pattern of most frequent confusions and omissions.

G23. Nonlinear time-scale modification of speech signal with varied segmental duration characteristics. Yoh'ichi Tohkura and Yoshinori Kitahara (ATR Auditory and Visual Perception Research Laboratories, Twin 21 MID Tower, 2-1-61 Shiromi, Higashi-ku, Osaka, 540 Japan)

Segmental duration of each phoneme changes depending upon the speaking rate. Generally, vowel parts are easier to be compressed or expanded than consonant parts are in fast or slow speech, respectively. Questions raised in this paper include how the speaking rate can be extracted from the speech signal without knowing the content (i.e., phonetic information) and what kind of time-scale modification can be chosen in order to control speaking rate. First, the segmental duration compressibility of the speech signal was defined by path slopes in DTW spectral matching when utterances with various kinds of speaking rates were matched to a reference utterance of a normal speaking rate. On the assumption that the compressibility is inversely proportional to segmental spectrum changes, the relationship between the compressibility and the average cepstral time difference Δ_{cep} [S. Furui, IEEE Trans. Acoust. Speech Signal Process. ASSP-34, 52–59 (1986)] was studied. The results showed that the Δ_{cep} is an efficient parameter to represent the compressibility. By using the Δ_{cep} as a control parameter of the speaking rate, nonlinear time-scale modification can be achieved without speech quality degradation.

Session H. Underwater Acoustics I: Signal Processing for Underwater Acoustics (Précis-Poster Session)

L. Neil Frazer, Cochairman
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Contributed Papers

Posters must be set up before 8:00 a.m. (before Opening Plenary Session). Following presentation of the précis, posters will be on display until 11:30 a.m.

8:30

H1. Comparison of spatial processing techniques for sonar array data. David V. Wyllie, Brian G. Ferguson, and Garry C. Speechley (Maritime Systems Division, WSRL, DSTO, Sydney, P. O. Box 706, Darlinghurst 2010, Australia)

It is demonstrated that adaptive beamforming techniques are superior to conventional methods for the spatial processing of sonar array data. For purposes of comparison, both conventional and adaptive frequency-wavenumber spectra are presented for various sonar arrays. Adaptive processing is found to improve the spatial resolution performance of an array of acoustic sensors and to suppress the spatial leakage inherent in conventional processing.

8:35

H2. Design of periodic signals for acoustic system identification using hard-limiting transducers. Murray D. Johns (Physics Department, University of Auckland, Auckland 1020, New Zealand)

Using periodic signals for underwater acoustic system identification enables signal averaging to be used to increase the signal-to-noise ratio of the received signal. The desired properties of such a signal are partially determined by the properties of the electroacoustic transducers used. This paper shows how to design signals to give the maximum receiver signal-to-noise ratio when the desired source signal is known but the source transducer has an ideal hard-limiting characteristic. In such cases, it is almost always better to generate a signal different from the desired one and to correct for the difference (using linear signal processing) in the receiver. Examples of such designs and the improvements to be gained will be given.

8:40

H3. The effect of target motion on bearing estimates from a beam interpolation algorithm. Robert A. LaTourette (Naval Underwater Systems Center, New London, CT 06320) and Lawrence C. Ng (Lawrence Livermore National Laboratory, University of California, Livermore, CA 94550)

An initial estimate of a target's bearing from an acoustic array of preformed beams is determined by choosing the beam pointing direction with the largest detected energy. An attempt to improve on this initial bearing estimate is accomplished by interpolating the response of three beams centered about the peak response. This interpolation process is designed to yield an improved estimate of the target bearing between beam pointing directions. This paper is a study of the merits of this beam interpolation process in the presence of target motion.

8:45

H4. Detection performance degradation due to signal uncertainties. Azizul H. Quazi and Albert H. Nuttall (Naval Underwater Systems Center, New London, CT 06320)

Detection of a known signal in white noise using a matched filter depends on the signal-to-noise ratio (signal energy to noise spectral density). As the uncertainties of signal characteristics, such as phase and amplitude, increase, the required SNR must be increased in order to maintain the same detection performance. However, in both cases of known and unknown signals, it is usually tacitly assumed that the signal covers the total bandwidth of the detector and the total integration time. But, in real ocean environments, it is difficult to know the exact signal arrival time and the exact bandwidth required to cover the wide range of Doppler shifts. As a result, the actual detector may utilize a wider bandwidth, and the integration time may extend beyond the signal duration. This paper addresses the problem where the signal duration L is less than or equal to the actual integration time T, and the signal bandwidth B is equal to or less than the actual bandwidth W of the detector. The performance loss is shown as a function of TW/BL as it varies over a range of 1 to 1000, for various combinations of probability of detection and probability of false alarm. [Work supported by NUSC special initiative program.]

8:50

H5. Detection performance of normalizer for multiple signals subject to partially correlated fading with chi-squared statistics. Albert H. Nuttall (Code 304, Naval Underwater Systems Center, New London, CT 06320)

The false alarm and detection probabilities for a multipulse signal subject to partially correlated fading, in the presence of Gaussian noise of unknown level, are derived in closed form. The number K of signal pulses, as well as the number L of noise-only pulses used to estimate the noise background power level, are arbitrary. The power fading is characterized by a chi-squared distribution with $2m$ degrees of freedom and a normalized set of covariance coefficients $\{\rho_{kl}\}$, all of which can be selected arbitrarily, in order to match an experimental realization or an actual measured situation. The performance capability of this processor depends additionally on the received signal-to-noise ratio. This study covers the case of a nonconstant threshold; comparisons of this normalizer with earlier results (for $L = \infty$) enable a quantitative evaluation of the losses incurred by lack of knowledge of the noise level. The important capability of constant false alarm rate is achieved by this normalizer.

8:55

H6. Directivity index of volumetric arrays using optimal amplitude shading. Robert Bruce Williams and M. Phillip Meacham (Naval Ocean Systems Center, Code 541, San Diego, CA 92152)

Anderson and Munson [J. Acoust. Soc. Am. **35**, 1162–1168 (1963)] showed that the directivity index (DI) of an infinitely densely populated spherical shell array was about equal to that of a sphere. Extrapolating to discrete elements, this means the shell requires far fewer elements. They did not compute DI using amplitude shading, due to the impractical cost of such systems at that time. Today's technology removes that constraint. This work revisits the problem with shading, using an approach for choosing the amplitude shading coefficents that maximizes DI [H.S.C. Wang, J. Acoust. Soc. Am. **57**, 1076–1084 (1975)]. Calculations have been made for the DI of shaded cubic volumetric arrays, forming beams perpendicular to one of its faces, in the presence of isotropic noise. Results show that for 27 and 125 element arrays with element matrix spacings of 1/2 wavelength, a full 10 log (number of elements) can be obtained for DI. Work is underway to investigate larger arrays and smaller spacings. The approach will also be extended to nonisotropic noise fields. [Work supported by NORDA and NOSC exploratory development programs.]

9:00

H7. Least-squares and single-filter always-convergent iterative deconvolution of transient signals for correlation processing. James H. Leclere, George E. Ioup,[a] Juliette W. Ioup,[a] and Robert L. Field (Code 244, NORDA, Stennis Space Center, MS 39529)

Correlation processing for distributed sensors is most accurate for short pulses and those whose autocorrelation is sharply spiked. For longer transient signals, multipath arrivals at each sensor have significant interference with each other, and it is difficult to identify individual arrival times. Deconvolution of the received signal to sharpen the transients is one method to decrease the overlap and increase the accuracy with which travel times can be identified. Deconvolution can also be applied after cross correlation to sharpen the autocorrelation of the transients. Least-squares deconvolution is the most commonly used approach for acoustic signals. It has the disadvantage of being computer intensive when filters for long transients are needed. An alternative approach, the single-filter application of the always-convergent iterative technique, is faster and provides variable control for noise. The two techniques are compared for actual underwater acoustic multipath transient signals. Single filter application of always-convergent iterative noise removal is compared to the use of a modified Blackman–Harris window for noise control. [a] Also at the Department of Physics, University of New Orleans.

9:05

H8. Comparison of double and triple cross correlation for arrival time identification of amplitude- and frequency-modulated acoustic transient signals. Juliette W. Ioup,[a] George E. Ioup,[a] Robert L. Field, and James H. Leclere (Code 244, NORDA, Stennis Space Center, MS 39529)

The triple cross correlation of three signals is a simultaneous function of two lags. It is an alternative to cross correlations taken two at a time for determining the lags for a given source at three distributed sensors. It should offer improvement in arrival time identification only when the statistics of the signal have significant third moment components. In this study, amplitude- and frequency-modulated snythetic transient signals are propagated over several possible paths to three sensors, and the triple correlation of the received pulses computed, as well as the cross correlations of the same three signals two at a time. The efficacy of these two approaches is compared for a variety of amplitude- and frequency-modulated transient signals and multipath interference conditions. [a] Also at the Department of Physics, University of New Orleans.

9:10

H9. *In situ* acoustic calibration for a large aperture array. Barbara J. Sotirin (Marine Physical Laboratory A-005, Scripps Institution of Oceanography, La Jolla, CA 92093)

During September 1987, a large aperture acoustic array was deployed vertically in the Northeast Pacific to study low-frequency noise in the ocean. Coherent combination of the 120-channel outputs requires knowledge of individual element amplitude and phase response for accurate results. Two *in situ* methods of array calibration are described and results from the September experiment are presented. The first method used transmissions from a low-frequency source of known location and power level. Simulating the conditions encountered during the transmission, the power arriving at the array was predicted by several acoustic propagation models. By comparing the array response at specific frequencies to the response predicted by the models, an absolute calibration was obtained. An error curve for the phase data was generated by unwrapping the phase, accounting for a sampling offset in the array, and subtracting a multiple linear regression curve. The second method determines relative amplitude levels by examining the average ambient noise power output of a specified frequency band across the array. Using spectral, coherence, and directionality plots, the level of self-noise in the array was shown to be below that of the ambient noise being measured. These two independent methods provide a consistent set of element calibration values used for array beamforming. [Work supported by ONT.]

9:15

H10. Abstract withdrawn.

9:20

H11. Matched-mode processing corrections for array tilt and bottom type. James A. Mercer (Applied Physics Laboratory, University of Washington, Seattle, WA 98105)

In a related effort, Homer Bucker has shown that matched-mode processing for an unknown sound-speed environment can be significantly improved if correction factors for the mode-line amplitude functions can be determined. The correction factors are obtained when a source with known location is available to calibrate the system. This paper describes the results of applying the same techniques for simulated cases of unknown array tilt and bottom characteristics.

9:25

H12. Self-consistent modeling of signal and noise in a three-dimensional environment. John S. Perkins, W. A. Kuperman, and F. Ingenito (U.S. Naval Research Laboratory, Code 5160, Washington, DC 20375-5000)

Previous propagation work is extended to model surface noise, shipping, and signal sources in a fully three-dimensional environment. The noise cross-spectral density matrix for a vertical array is computed as the sum of a local contribution and propagation from distant small patches of ocean surface. Propagation from any point to the array is made efficient by precalculating and storing the local normal modes for every distinct path. The usual cylindrical coordinate system for the propagation calculation is obtained, and various sound sources are added at any location and are propagated to the array. For example, a storm is simulated by increasing the source strength of the appropriate patches of surface. Results demonstrate that the vertical directionality on the array reflects the range-dependent environment.

9:30

H13. Environmental symmetry breaking: An application of 3-D matched-field processing. John S. Perkins, W. A. Kuperman, and F. Ingenito (U. S. Naval Research Laboratory, Code 5160, Washington, DC 20375-5000)

Self-consistent simulations of matched-field processing in a noisy three-dimensional environment are produced by using the same numerical model for constructing the signal and correlated noise field. In principle, matched-field processing in an asymmetric three-dimensional (range- and azimuthal-dependent) ocean provides the possibility of horizontal beamforming with a purely vertical array since the environment itself breaks the azimuthal symmetry of the vertical array. This scheme is called environmental symmetry breaking (ESB). As a demonstration, an ocean environment is used that has a portion of the Gulf Stream running through the area of interest together with a storm that produces a correlated horizontally (as well as vertically) anisotropic noise field. Linear and nonlinear matched-field processing is considered for a vertical array. These simulations provide the first numerical evidence that ESB can be applied to the underwater detection problem.

9:35

H14. Sensitivity of matched-field processing to sound-speed profile mismatch for vertical arrays in a deep water Pacific environment. A. Tolstoy (U. S. Naval Research Laboratory, Washington, DC 20375-5000)

In this paper, the sensitivity of matched-field processing (MFP) to sound-speed profile mismatch (based upon archival profiles resulting in various degrees of mismatch) will be examined with emphasis on sources within 50 km. A 10-Hz source will be considered whose field is generated by a normal mode model where only the water-borne energy is used, thereby eliminating issues relating to the estimation of bottom parameters. Array parameters, i.e., number of phones and array depth, will be examined for their effects on range and depth localization for various degrees of mismatch. In particular, it will be indicated where an array is most and least sensitive to sound-speed mismatch as a function of depth from the surface and range from the source. Also, the issue of selecting the range increment for the minimum variance (Capon) processor will be briefly addressed and it will be pointed out how too crude an increment can lead to erroneous results. Finally, an example will be presented of coherent broadband (BB) MFP illustrating improved robustness and sidelobe suppression relative to both single frequency and incoherent BB results.

9:40

H15. A comparison of matched-mode and matched-field processor performance against sound-speed and sensor position mismatch. Ellen S. Livingston (Naval Research Laboratory, Code 5120, Washington, DC 20375)

Recent results due to T. C. Yang [J. Acoust. Soc. Am. **82**, 1736–1745 (1987)] have shown that matched-mode processing can provide range-depth source localization with good sidelobe and array gain performance. His simulations and success with limited data suggest that the matched-mode processor is relatively robust with respect to environmental and hydrophone position mismatch. The objective of this study is to compare the performance of matched-mode versus matched-field processing under identical array geometry and environmental conditions. Assuming a range-independent deep ocean environment, sound-speed mismatch and sensor location mismatch were simulated at second convergence zone ranges. Eigenvalue truncated (reduced rank inverse) matched-mode processor performance was compared to conventional matched-field processor performance at 30 Hz on a large aperture vertical array. In the absence of mismatch, the two methods yield near identical results on an array spanning half the water column. The results in the presence of mismatch will be discussed and quantified in terms of signal gain degradation, sidelobe level, and speckle.

9:45

H16. A new efficient method of source localization in waveguides processing in mode space. E. C. Shang (NOAA/Wave Propagation Laboratory, Boulder, CO 80303)

Matched-field processing can either be performed in phone space (matching the total field received by each phone) or in mode space (matching the resolved modes). For a stratified waveguide, more efficiency can be obtained in mode space due to the fact that: (1) the source range information and depth information are separable in mode space and the number of replicas is tremendously reduced from $(N*M)$ to $(N+M)$, where N is the range gridding number and M is the depth gridding number, and (2) more confidence can be obtained by making a choice from the resolved modes. In this paper, the "modal feature matching" (MFM) technique has been proposed instead of the "modal beamforming" (MB) technique. One of the advantages of this method is that high performance can be achieved even if only a few modes are available, especially for depth estimation, when two modes are sufficient. Numerical examples are presented.

9:50

H17. Modal amplitude filtering of vertical array data in a shallow water environment. George M. Frichter, IV (Naval Ocean Research and Development Activity, Stennis Space Center, MS 39529-5004)

Current research at NORDA into new matched-field processing techniques based on nonlinear matching of complex modal amplitudes filtered from noisy vertical array data shows that very low SNR sources may be detected and localized when spatially correlated (modal) noise dominates. All mode-matching algorithms require knowledge of the local ocean environment to calculate depth eigenfunctions for the waveguide. This modal structure is then incorporated into a mode filter that deduces complex modal amplitudes from the complex acoustic pressures measured on the array. However, discretely sampling a noisy incident acoustic field means that the modal amplitudes will not be recovered exactly. These modal amplitude errors can cause degradation of both detection and localization performance of modal matched-field processors. In this study, three techniques for mode filtering are examined from the point of view of processor detection degradation. Computer simulations demonstrate how array configuration and noise combine to produce mode filter errors and how these errors translate into processor degradation. In addition, an analytic description of the relation between mode filter error and processor performance is discussed.

9:55

H18. Modal beamforming array gain. T. C. Yang (Naval Research Laboratory, Code 5123, Washington, DC 20375)

In a previous paper [T. C. Yang, J. Acoust. Soc. Am. **82**, 1736–1745 (1987)], range and depth were successfully estimated by modal beamforming. In this paper, the modal beamforming array gain in closed form for a vertical array of N hydrophones is derived. For the case of white noise and a long well-populated vertical array, the array gain nearly equals 10 log N. For the case of colored noise and a finite aperture, finite element vertical array, the array gain degrades from the theoretical by an amount determined by the mode-depth amplitudes evaluated at the source and receiver depths, and the modal covariance matrix of the noise. Numerical estimates of array gain are given for the FRAM IV vertical array for a colored noise environment in the Arctic sound channel. Approximately 1-dB array gain degradation is found at 47 Hz. Proper array configuration and deployment depth is not only important for optimizing array gain but also critical for maximizing the input signal-to-noise ratio on a vertical array, which together with the array gain determines the array output signal-to-noise ratio.

10:00

H19. Modal beamforming in a range-dependent environment. T. C. Yang (Naval Research Laboratory, Code 5123, Washington, DC 20375)

Modal beamforming, previously developed for a range-independent case [T. C. Yang, J. Acoust. Soc. Am. **82**, 1736–1745 (1987)], is extended to a range-dependent environment. The range-dependent sound channel is partitioned into consecutive waveguides as a function of range. The normal modes in each waveguide are determined by the sound-speed and bottom profile in that waveguide. The mode coefficients in adjacent waveguides are related by matching the boundary condition at the vertical interface. Modal beamforming is conducted for each waveguide. The array outputs from the waveguides are combined and plotted as a two-dimensional surface with respect to range and depth. The source location is determined by the peak of this surface. A numerical example is given based on simulated data for a source propagating through a series of different sound-speed profiles. The simulated data are created using an improved coupled-mode program that was initially developed by R. B. Evans. Range and depth are estimated using a long vertical array of 3000-m aperture, containing 50 phones. Modal beamforming for a range-dependent environment can also be conducted directly using the coupled-mode program or any other propagation code.

10:05

H20. Self-coherent matched-field processing. H. P. Bucker (Naval Ocean Systems Center, San Diego, CA 92152)

In this paper, a simple self-cohering procedure is proposed to correct for unknown environmental parameters and inaccurate model inputs. Consider matched-field processing for one or more vertical line arrays and assume that range changes are not strong, or abrupt, so that an adiabatic mode propagation model can be used. The standard (Bartlett) estimator of a sound source being at a particular location [$\mathbf{r} = (x,y,z)$] in the ocean can be written as

$$B(\mathbf{r}) = \sum_k \sum_j p_{kj}^* \sum_m \phi_m(0,z_{kj})\phi_m(r,z) \exp\left(i \int_0^r w_m \, dr\right),$$

where p_{kj} is the Fourier coefficient, in a small frequency bin, of the pressure received at sensor j of the kth vertical line array, ϕ_m is the depth response of mode m, w_m is the horizontal wavenumber of mode m, and * denotes the complex conjugate. It has been shown [e.g., T. C. Yang, J. Acoust. Soc. Am. **82**, 1736–1745 (1987)] that modal decomposition is an efficient means to perform MFP. Define U_{mk}, the mode-line response function, as the correlation between the depth function of mode m with the complex signals received at vertical line k; that is,

$$U_{mk}^* = \sum_j p_{kj}^* \phi_m(0,z_{kj}).$$

Now we can write

$$B(\mathbf{r}) = \sum_k \sum_m U_{mk}^* \phi_m(r,z) \exp\left(i \int_0^r w_m \, dr\right).$$

There will always be differences between sound propagation in environmental models and in the "real"ocean due to unknown mesoscale processes and inaccurate model input parameters. If at some location in the ocean there is a known acoustic source (which could be a surface ship), a set of mode-line correction factors Q_{mk}, which are complex numbers that "correct" the U_{mk}, can be calculated to give maximum correlation at the particular location of the known source. The important question is how useful are these correction factors at points (in the 5-D space of range, bearing, depth, frequency, and time) away from the point of the known source? This paper describes some preliminary calculations to determine the usefulness of this self-cohering method.

TUESDAY MORNING, 15 NOVEMBER 1988 HONOLULU/KAHUKU ROOM, 9:30 TO 11:54 A.M.

Session I. Architectural Acoustics II and Musical Acoustics I: Concert Acoustics: The Performers' Perspective

R. Lawrence Kirkegaard, Cochairman
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Shun-ichi Nakamura, Cochairman
Tokyo National University of Fine Arts and Music
12-8 Ueno-koen, Taito-ku
Tokyo, 110 Japan

Chairman's Introduction—9:30

Invited Papers

9:35

I1. Acoustic conditions preferred by performers. Ichiro Nakayama (The Institute of Scientific and Industrial Research, Osaka University, 8-1 Mihogaoka, Ibaraki, 567 Japan)

To obtain a guideline for the stage enclosure design in a concert hall for a performer, subjective preference judgments for synthetic music sound fields have been performed by alto-recorder soloists in relation to objective parameters. The sound fields consisted of two early reflections and their subsequent reverberation with systematically varied amplitudes, delay times, reverberation times, angles of incidence, and music used as source signals. Music motifs were characterized by their coherence of autocorrelation function (ACF). It was found that the most preferred delay and reverberation times could be determined by the ACF and the amplitude of the reflections. The results obtained here suggest the importance of a movable wall in the stage enclosure, adjusting the distance from the performer, and suggest that such a wall should be taken into consideration in the stage enclosure design.

9:55

I2. Concert acoustics: The performers' perspective. R. Lawrence Kirkegaard, Robert F. Mahoney, and Dana L. Kirkegaard (Kirkegaard & Associates, 4910 Main Street, Downers Grove, IL 60515)

Following each performance during the San Francisco Symphony's recent tours of Europe, Asia, and North America, musicians completed questionnaires evaluating the stage acoutics of the hall in which they had just performed. The questionnaire's format was developed with the input of the Symphony and in deference to an earlier study by A. C. Gade [A. C. Gade, "Musicians' Ideas About Room Acoustical Qualities," Rep. No. 31, Acoustics Lab., Techn. Univ. of Denmark (1982)]. Subjective parameters included the ability to hear each section of the orchestra (including one's own), the ease of ensemble, the ease of tone production, adequacy of dynamic range, loudness, and the support and responsiveness of the room. These data were compared with echograms recorded at five locations, i.e., 25 communication paths, on each stage. Interpretations of these data have been compared with the published results of research with laboratory-simulated stage acoustics and are presented in this series of papers addressing loudness, communication, and support and responsiveness.

10:15

I3. Acoustic components supporting solo singers on concert hall stages. Shun-ichi Nakamura (Tokyo National University of Fine Arts and Music, 12-8 Uenokoen, Taito-ku, Tokyo, 110 Japan) and Shin-ichiro Kan (Kajima Institute of Construction Technology, 2-19-1 Tobitakyu, Chofu, 182 Japan)

When singing on stage, soloists require their surroundings to have certain reflective properties such that they feel acoustically supported by the hall. Soloists feel this support in some cases, but not always. It has been found that, in simulated environments, subjects can receive the impression that they are in a good concert hall, even with only a single reflection. However, the subjects' perceptions of the degree of frontal reflection were greater than the actual amount of such reflection. Various acoustic components, including direct feedback of the subjects' sound, "couple" together to amplify the perceptible frontal reflection. Primary reflections from the stage enclosure seem to be not so effective by themselves, but the stage's local reverberant field is important, particularly in large halls, in that it couples with the relatively weak frontal reflection from the auditorium. It has also been found that a forward masking effect against reverberation can occur. Results from scale model experiments suggest that there are simple size effects directly related to the level and to the decay rate of reverberant sound. It was proposed that the stationary level and the early decay rate of the reverberant sound field, as well as the rate of the reverberation's frontal component, at a certain specified time should be measured *in situ* on the stage in order to get more information regarding acoustic engineering.

10:35

I4. Playing with hornists. Edward McCue (Wenger Corporation, 555 Park Drive, Owatonna, MN 55060)

An examination of the acoustical requirements of the orchestral horn section during rehearsal and performance serves as a point of departure for the design of performance platforms and orchestra pits. In addition to displaying an asymmetrical radiation pattern that is disadvantageously oriented to the right and rear of the player, the spectrum of the horn is easily masked by other sections of the orchestra. The instrument's timbre and physical placement within the orchestra are reviewed with respect to the historical evolution of orchestration and the architectural design of opera houses and concert halls. Contemporary difficulties with excessive loudness and ensemble are discussed. Architectural design criteria leading to a good performance environment for the horn section are suggested.

Contributed Papers

10:55

I5. The acoustic design of partially enclosed orchestra pits. Robin K. Mackenzie (Department of Building, Heriot-Watt University, Edinburgh EH14 4AS, Scotland, United Kingdom)

The trend during recent years for musical directors of touring opera companies to require productions to be played to a fuller score has resulted in problems for the orchestral musician. Research has found that musicians frequently complain of (a) lack of contact with the stage, (b) diminished ensemble within the orchestra, and (c) lack of feedback from the auditorium. The geometric design of the orchestra pit has been investigated and optimum parameters have been established. A system of electroacoustic reinforcement to improve communication between the stage and pit is described. [Work supported by S.E.R.C.]

11:07

I6. Can ensemble condition in a hall be improved and measured? A. C. Gade (Acoustics Laboratory, Technical University of Denmark, Lyngby, Denmark)

In collaboration with the Danish Broadcasting Corporation an extensive series of experiments has been carried out in The Danish Radio Concert Hall with the practical purpose of trying to improve the ensemble conditions on the platform for the resident symphony orchestra. First, a series of experiments in a 1:20 scale model indicated that among several suggested means the following would be the most effective and acceptable: (a) changing the shape of the sidewalls in the platform area in order to make them reflect sound back to the musicians more effectively; (b) lowering and redesigning of the ceiling reflectors; and (c) changing the position of the orchestra on the platform. These variables were then tested in full scale experiments in the hall including subjective evaluation by the orchestra in order to verify their effects under practical conditions. New objective parameters, which showed very high correlations with the subjective data, also made it possible to compare the improvements with conditions as recently measured in famous European Halls. Besides providing the needed results, the experiments also shed some light on how musicians change their criteria for judging acoustic quality depending on the experimental situation—a fact which had become evident from our previous research on musicians' room acoustic conditions.

11:19–11:24

Break

11:24–11:54

Panel Discussion

TUESDAY MORNING, 15 NOVEMBER 1988
KAUAI ROOM (WEST END), 10:00 A.M. TO 12:00 NOON

Session J. Speech Communication II: Analysis and Synthesis, Part B (Poster Session)

James Hillenbrand, Cochairman
RIT Research Corporation
Rochester Institute of Technology
75 Highpower Road
Rochester, New York 14623

Hirokazu Sato, Cochairman
NTT Human Interface Laboratories
3-9-11 Midori-cho
Musashino, 180 Japan

Contributed Papers

Posters should be set up before 10:00 a.m. All posters will be displayed from 10:00 a.m. to 12:00 noon. To allow contributors the opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 10:00 to 11:00 a.m. and contributors of even-numbered papers will be at their posters from 11:00 a.m. to 12:00 noon.

J1. Formant extraction by local approximation of the speech spectrum. Kensaku Fujii and Juro Ohga (Communication and Space Division, Fujitsu Laboratories, Nakahara-ku, Kawasaki, 211 Japan)

The present study proposes a new method of formant extraction that flattens the residual signal with reduced computational load. In this method, the formants are extracted in reducing order of the amplitude. Each formant is represented by an IIR filter of

$$F(Z) = (1 - 2Q_a \cos\theta_0 Z^{-1} + Q_a^2 Z^{-2})/$$
$$(1 - 2Q_b \cos\theta_0 Z^{-1} + Q_b^2 Z^{-2}).$$

This formula has three unknown quantities Q_a, $\cos\theta_0$, and Q_b. They are estimated from quadratic equations on the speech spectrum. The coefficient $\cos\theta_0$ corresponds to the formant frequency, which is obtained as the peak of the second-order curve defined by the maximum line spectrum A and the neighboring line spectra B and C. Here, Q_b is related to the formant bandwidth which is given by the geometrical series average

$$1/(1 - Q_b) = [1/(1 - e^{aT}) + 1/(1 - e^{bT})]/2.$$

Here, e^{aT} and e^{bT} are the damping terms in the impulse response of the second-order IIR filter. The relative magnitude of the frequency responses of this filter at the frequencies of the line spectra A and B is set to coincide with the relative amplitude of the line spectra A and B. The e^{bT} is determined in a similar way by using the line spectra A and C. The coefficient Q_a is the damping term of the frequency response which expresses the global shape of the spectrum. It can be estimated by using the line spectrum A and the maximum frequency component.

J2. Formant-frequency estimation by linear transformation of the LPC-cepstrum. David J. Broad (1627 Bath Street, Apt. 3, Santa Barbara, CA 93101) and Frantz Clermont (Computer Sciences Laboratory, Australian National University, Canberra, ACT 2601, Australia)

Correlations near 0.98 between vowel formant frequencies and linear combinations of 1/3-octave-band spectrum levels reported by Pols *et al.* ["Perceptual and physical space of vowel sounds," J. Acoust. Soc. Am. **46**, 458–467 (1969)] suggest that the formants might be estimated by linear transformation of either a low-resolution log spectrum or of the first M cepstral coefficients c_i, which are linear functions of a log spectrum. Applying this idea with $M = 14$ to four speakers results in pooled intraspeaker prediction errors of 41, 126, and 134 Hz for F_1, F_2, and F_3, respectively. These become 52, 164, and 218 Hz when regressions on data from three speakers are used to measure formants from a fourth. The method is therefore not very accurate, but it is robust in that large errors from misidentified peaks are rare. A study of single-resonance waveforms explains why the method works: The c_i are roughly cosines in the resonance frequency F, with i half-cycles over the analysis bandwidth; these $c_i(F)$ form a basis set for approximating the function $F_{est} = F_{real}$.

J3. Formant contour extraction by a temporally constrained search of the spectral resonance space. Frantz Clermont (Computer Sciences Laboratory, Research School of Physical Sciences, Australian National University, GPO Box 4, Canberra, ACT 2601, Australia)

An algorithm for extracting the first three formant-frequency contours of vowel sounds is based on properties of the linear prediction phase spectrum derived by Yegnanarayana [J. Acoust. Soc. Am. **63**, 1638–1640 (1978)], and Yegnanarayana and Reddy [Int. Conf. Acoust. Speech Signal Process., Conf. Record, 744–747 (1979)]. The former study has shown that the negative derivative of the linear prediction phase spectrum (NDPS) behaves like a formant-enhancing filter. The latter study has shown that the Euclidean distance between pairs of NDPS emphasizes differences only around formant peak regions, and that it is easily computed as an index-weighted cepstral distance. In stage 1 of the proposed algorithm, all posible sets of four candidate peaks from the original spectrum are used to synthesize four-formant spectra. The index-weighted cepstral distances between these simplified spectra and the original spectrum become row entries in a matrix of intraframe distances. In stage 2, a dynamic programming (DP) procedure imposes continuity constraints across frame spectra. The DP-cost function is the cumulative sum of the intraframe distances plus the minimum of interframe cepstral distances. Extracting temporally constrained contours then consists of backtracking the optimum path through the spectral distance matrix.

J4. Statistical approaches to formant tracking. Robert T. Gayvert and James Hillenbrand (RIT Research Corporation and Rochester Institute of Technology, 75 Highpower Road, Rochester, NY 14623)

Formant trackers that rely on peak picking tend to make occasional large errors. This paper investigates two general methods for determining formant locations without explicit use of peak information. Both approaches involve statistical models derived from hand-traced formant tracks. In the first method, individual formant frequency values are estimated using a maximum likelihood classifier. In the second, formant probability distributions are found for each element of a vector quantization codebook, and formant values are then determined by conditional mean estimates. Hidden Markov models or simple smoothing can then be applied to provide continuity constraints. Both of these techniques have been quantitatively analyzed using a database of 78 utterances produced by four males and four females. The performance of these trackers across different training and testing sets will be discussed. [Work supported by Rome Air Development Center under contract F3060285-C-0008.]

J5. A pitch extraction method using higher-order joint moment. Norio Nomura and Yuichi Yoshida (Department of Electrical and Electronics Engineering, Sophia University, 7-1 Kioicho, Chiyoda-ku, Tokyo, 102 Japan)

A modified higher-order joint moment method for a correct pitch extraction is proposed. The mean products of three vectorized samples are used, while the usual autocorrelation method uses the mean products of two samples. Each of three samples separated by the same sample interval is modified to a vector whose modulus is equal to the absolute value of the sample, and the argument is $\pm$ 120 deg in accordance with the sample's sign. Only when three original samples have the same sign does the product of three vectors become a real positive value. By averaging the products in a certain window, a sharp peak on the sample-interval axis is obtained at the pitch period. The effectiveness of the method is examined with real voiced signals. The peak at the pitch period is much sharper than the peak of the autocorrelation method, and the spurious peaks are very small. These properties result in a correct pitch extraction. It is also shown that the influence of the pitch fluctuation has properties similar to those as in the autocorrelation method.

J6. Formant frequency estimation by moment calculation of the speech spectrum. Kazuyuki Takagi and Shuichi Itahashi (Institute of Information Sciences and Electronics, University of Tsukuba, Tsukuba, 305 Japan)

Moment calculation is applied to extract the formant frequencies of a speech spectrum. Three kinds of first-order moments divide a spectrum into four frequency regions. The centers of gravity of the first three regions are calculated to give the 0th order estimation of the 1st, 2nd, and 3rd formant frequencies. Then the upper and the lower bounds of each region are modified so that the estimated frequency comes closer to the major peak of the spectrum, utilizing the second-order and the third-order moments that represent the variance and skewness of the spectral pattern. The process repeats until the k th estimation equals the $(k-1)$th estimation. This modification improves the estimation precision significantly. An experiment with model spectra generated by an all-pole model gave estimation precision of 3% using formant frequencies typical of the five Japanese vowels. Speech materials spoken by five male and five female speakers were used for this experiment. The speech waveform was sampled at a rate of 10 kHz through a 5 kHz LPF, quantized into 12 bits; then the spectrum envelope was calculated with the first 24 cepstra of a 256-point FFT spectrum. The results give acceptable precision, compared with visually determined formant frequencies.

J7. Real-time pitch detection with a digital signal processor. Michiharu Mito, Kiyoshi Takahasi, Syuji Kurokawa, Syogo Nakamura, and Tadahiro Kubota (Department of Electrical Engineering, Tokyo Denki University Chiyoda-ku, Tokyo, 101 Japan)

Pitch detection is an important and essential technique in speech analysis, synthesis, and so on. There are many methods to extract voice pitch, but it is difficult to perform pitch extraction in real time. This paper describes a simple real-time pitch detection algorithm which directly estimates the interval between peaks of the waveform. This algorithm consists of the following two parts: one is the peak emphasis of voiced signals and the other is the pitch detection. The peak emphasis is obtained by running a DFS (discrete Fourier series) and a window operation. It is important to determine the peaks for the pitch measurement because the pitch period is obtained by estimating the interval between successive peaks representing the pitch period of the voiced signal. The peaks related to pitch are determined using a few simple rules. Since the voiced signal waveform includes several extra complicated peaks, these rules are constructed taking into account the characteristics of the voiced signal. Other peaks, which do not correspond to the pitch period, are rejected by a simple logical judgment. A real-time pitch detection algorithm has been realized using a conventional DSP.

J8. Speech analysis using a time-varying ARX model for separating the source-tract coupling of vowels. Tetsuo Funada (Faculty of Engineering, Kanazawa University, 2-40-20 Kodatsuno, Kanazawa, 920 Japan)

The purpose of this research is to extract formant frequencies precisely and to classify voiced/unvoiced intervals accurately based on a source-tract model. A sequential estimation of the source wave (i.e., the glottal volume flow) and the vocal tract (VT) characteristics is achieved by using a time-varying "ARX model," where the term ARX model refers to an AR (autoregressive) model with an auxiliary nonwhite input (X input). This X input indicates the glottal volume flow in the present research. Applications to synthetic vowels generated by the two-mass model demonstrated the following results: (1) Much information on the glottal closure and opening was obtained from the X input; and (2) compared to the conventional (autocorrelation) LP method, formant frequencies (especially the first formant) during the open period of the glottis were estimated more accurately. It has also been observed from real vowels uttered by a male speaker that the phase of the X input agrees with the phase of the glottal movement which can be confirmed by electroglottography (EGG).

J9. Excitation problem in speech synthesis. Bishnu S. Atal (Acoustics Research Department, AT&T Bell Laboratories, Murray Hill, NJ 07974)

Linear predictive coding methods require efficient representation of both the LPC filter and its input excitation to synthesize high-quality speech at low bit rates. Considerable progress has been made so far in encoding the filter parameters and it is possible to quantize these parameters with only 1600 bits/s without introducing distortion in the synthetic speech signal. However, it is still not possible to encode the LPC filter excitation at low bit rates and maintain high voice quality in the synthetic speech signal. In this paper, the problems associated with low bit repre-

sentation of the excitation are discussed. To achieve low bit rates, a parametric representation is needed that can provide a compact yet accurate representation of the excitation. Such a compact representation is obtained by expressing the excitation waveform as a linear combination of the eigenvectors of the autocorrelation matrix of the LPC filter's impulse response. This representation allows the study of the effect of changes in the filter excitation on the speech output in a systematic manner. The signal-to-noise ratios necessary to represent various eigenvector components in the excitation without producing perceptible distortion in the output speech signal have been determined. Thus the minimum number of bits necessary to reproduce a speech signal is estimated. These results will be discussed in the paper.

J10. Effects of fundamental frequency contour on the identification of resynthesized vowels with static formant frequency patterns. James Hillenbrand and Robert T. Gayvert (RIT Research Corporation and Rochester Institute of Technology, 75 Highpower Road, Rochester NY 14623)

At a previous meeting, a study that was aimed at determining the identifiability of vowels based exclusively on static spectral information was discussed [Hillenbrand and McMahon, J. Acoust. Soc. Am. Suppl. 1 **82**, S81 (1987)]. In that study, a formant synthesizer was used to generate steady-state versions of 1520 vowels (76 speakers × 10 vowels × 2 repetitions) using Peterson and Barney's measured values of $F0$ and $F1$–$F3$ [J. Acoust. Soc. Am. **24**, 175–184 (1952)]. The values of *all* control parameters remained constant throughout the 300-ms duration of each stimulus. Listeners in that study showed an error rate of approximately 25%, several times greater than the 5.6% error rate reported in the original Peterson and Barney study. The present study represents a follow-up designed to determine what role fundamental frequency movement might play in vowel identification. The new stimuli were identical to those of the previous resynthesis study except that all stimuli were generated with a falling pitch contour. Preliminary results suggest that the introduction of pitch movement decreases the error rate from approximately 25% to approximately 21%. [Work supported by Rome Air Development Center under contract F3060285-C-0008.]

J11. Analysis and synthesis of CV syllables in Hindi. S. S. Agrawal (Central Electronics Engineering Research Institute Centre, CSIR Complex, Hillside Road, New Delhi 110012, India)

Twenty-nine consonants of frequent occurrence in Hindi were combined with a cardinal vowel /ɑ/ to make CV-type syllables. These were spoken by a standard male speaker. The spectral analysis was done using a sound spectrograph and the covariance method of LPC analysis. The formant frequencies and their bandwidths were obtained in segments of varying duration from 8 to 20 ms depending upon the nature of acoustic information in the syllable. A program called "SNDSYS" was used to estimate the fundamental frequency and overall amplitude values at every 10-ms duration. These parameters were used to give a basic acoustic description and frame rules for consonant–vowel combinations. A P.C. version of Klatt's synthesizer called "KLPC" was used to synthesize the CV syllables. There are over 40 parameters and constants that are used in the default synthesizer configuration. Based on the acoustic description of different sounds, various parameters were updated at an interval of 5 to 10 ms to generate different consonant–vowel combinations. The configuration file obtained for each syllable (named as documentation file) was further used to update and change the parameters to improve the quality of synthesized speech. Special considerations related to the synthesis of voiced and unvoiced aspirated consonants of Hindi are also discussed.

J12. Synthesis of Chinese by rules based on a multipulse excitation model. Li Changli and Mo Fuyuan (Institute of Acoustics, Academia Sinica, Beijing, People's Republic of China)

According to the model of speech production, the characteristic parameter of speech can be divided into two parts: excitation and vocal tract parameters. Atal proposed the multipulse excitation model that can produce high-quality sythesized speech. This research shows that the intensity, duration, and pitch mode of single syllable Chinese produced by multipulse excitation may be changed when the adaptive method is utilized to process its multipulse sequences and vocal tract parameter. There are about 10 000 Chinese words in common use, but the pronunciation of many words is the same, so that only about 1300 syllables are independent. The Chinese language is a tone language. Each Chinese word is of four pitch modes, and the vocal tract parameter for the four modes of one word is almost the same. Therefore, there are 400 independent vocal tract parameters and 1300 multipulse sequences in Chinese. Based on the above strategy, a new method of sythesizing Chinese by rules has been proposed. The intelligibility and naturalness of the synthesized speech are satisfactory.

J13. Statistical modeling of dynamic spectral patterns for a speech synthesizer. Satoshi Takahashi, Yasuaki Satoh, Takeshi Ohno, and Katsuhiko Shirai (Department of Electrical Engineering, Waseda University, 3-4-1 Ohkubo, Shinjuku-ku, Tokyo, 160 Japan)

A new method called the spectral locus control method (SLCM) is proposed, which can approximate the dynamic characteristics of the speech spectrum, such as in the transition from vowels to consonants or from consonants to vowels, effectively and accurately. The main procedures of the method are as follows. Continuous speech is segmented into VCV units, and these units are grouped according to the consonants. The spectrum patterns of the V1CV2 units in each group are analyzed to construct a statistical model which, given the spectra of V1 and V2, generates the spectrum loci for V1CV2 units. To synthesize continuous speech, a spectrum appropriate for a given consonantal context is first selected for each vowel V in every CVC sequence in the text. Then, the temporal sequence of the spectrum patterns for the entire V1CV2 is calculated based on the spectrum of the stationary parts in V1 and V2. Since VCV segment spectra are adapted to their consonantal environment, the synthesized speech is highly natural, especially in transitions.

J14. A speech synthesis system by rule in Japanese. Ryunen Teranishi (Department of Acoustic Design, Kyushu Institute of Design, Shiobaru, Minami-ku, Fukuoka, 815 Japan)

In Japanese, text-to-speech systems have to deal with problems of complicated orthography and a writing custom without a clear word separation rule. The system shown here is a tentative one avoiding such troublesome problems, constructed to study the rules in Japanese that are useful for the speech synthesis. In order to obtain natural prosody, as in human text reading, the system should have some mechanism that divides the input sentence into several proper breath groups with pauses based on the analysis of the syntactic structure as a human does. The construction of the system has been accomplished, and it can respond to such a demand. This algorithm was realized as completed software on 2HD diskette, available for the PC-98 series of personal computer made by NEC. The features of the system are as follows. The input form is word units written in *kana* letter codes. These units are separated with the space code. No prosody code is necessary except for the punctuation marks. An original parser, based on traditional Japanese grammar, produces the prosody for the synthesized speech.

J15. A system for speech synthesis from Japanese orthographic text. Hisashi Kawai, Keikichi Hirose, and Hiroya Fujisaki (Department of Electronic Engineering, Faculty of Engineering, University of Tokyo, Bunkyo-ku, Tokyo, 113 Japan)

A system has been developed for speech synthesis from Japanese orthographic text of Japanese. The system consists of four processing stages. The linguistic processing stage utilizes natural language processing techniques for extracting lexical, syntactic, semantic, and discourse information from each paragraph of the input text. The phonetic processing stage utilizes this information to derive a string of segmental and prosodic symbols for the entire paragraph. The acoustic processing stage generates time-varying patterns of parameters from these symbols to control the final stage, which is a formant-type synthesizer. The Fujisaki–Ljungqvist model is adopted for the excitation of the voiced sounds [Proc. ICASSP

86, 1605–1608 (1986)], and its fundamental frequency is controlled by a model of F_0 contour generation [H. Fujisaki and K. Hirose, J. Acoust. Soc. Jpn. (E) 5, 233–242 (1984)]. The segmental features, on the other hand, are synthesized by concatenating pole-zero frequency patterns prestored for each syllable. The validity of the system, especially of the prosodic feature synthesis, was confirmed by the naturalness of the accent and intonation of the synthesized speech. [Work supported by Grant-in-Aid for Scientific Research on Priority Areas from Ministry of Education, Science and Culture of Japan, No. 63608002.]

J16. Text-to-speech system for English and Japanese. Kenji Matsui, Noriyo Hara, Masaaki Kitano (Central Research Laboratory, Matsushita Electric, 3-15 Yakumo Nakamachi, Moriguchi, Osaka, 570 Japan), Hector Javkin, Kazue Hata, and Hisashi Wakita (Speech Technology Laboratory, 3888 State Street, Santa Barbara, CA 93105)

A real-time text-to-speech system for English and Japanese has been developed. This system consists of a language processing module, a phonetic acoustic processing module, and a synthesis module. Full general English and Japanese sentences can be converted to speech. The Japanese software and English software are independent except for the synthesis module. The features of this system are as follows. (1) The synthesis module is a phoneme-based cascade-parallel formant synthesizer with high observed intelligibility (73.5% for the 119 Japanese monosyllables). (2) This system has a 3000-morphene English dictionary and 40 000-word Japanese dictionary with a high-speed search algorithm. (3) A large speech database was collected for the development of Japanese prosody rules. (4) For the precise control of pitch contour, the Fujisaki model was adopted. (5) One of the two systems developed can stand alone; the other requires a personal computer with a high-speed DSP board. (6) In the development of this system, some powerful interactive tools have also been developed for varying speech parameters in real time.

J17. The Japanese speech synthesis system with text editing and automatic prosodic control facilities. Seiichi Yamamoto, Norio Higuchi, and Tohru Shimizu (KDD Kamifukuoka R&D Laboratories, 2-1-15 Ohara, Kamifukuoka, 356 Japan)

A Japanese speech synthesis system using a Japanese speech synthesizer by rule and software for text editing and automatic prosodic control has been developed. In the case of Japanese, there is a possibility that difference in accent type will cause differences in meaning, and that a word included in a compound word, such as a compound noun or verb, will have an accent type different from its original one. Therefore, the correct specification of the prosodic symbols is not so easy, even for people without any difficulty in hearing. The software for the text editing and for the automatic prosodic control inserts symbols for the prosodic control into the input text during the process of the text creation and editing in which the input *kana* string is converted to a *kanji-kana* mixed sentence as in conventional Japanese word processors. On the other hand, the speech synthesizer uses phonemes as synthesis units and generates all acoustic parameters based on the 156 feature rules and 472 parameter rules. Since it also has the facility to send the synthetic speech through the telephone line, even a speech-impaired person can transmit messages to a distant listener.

J18. A Japanese speech synthesizer based on production rules. Norio Higuchi, Seiichi Yamamoto, and Tohru Shimizu (KDD Kamifukuoka R&D Laboratories, 2-1-15 Ohara, Kamifukuoka, 356 Japan)

A Japanese speech synthesizer by rule, which uses phonemes as synthesis units and generates all acoustic parameters based on production rules, has been developed. The conversion from the input *romaji* string in Hepburn style to the synthetic speech waveform consists of (1) the generation of the phoneme/boundary string with the distinctive feature matrix based on 156 feature rules, (2) the conversion to sequences of the acoustic parameters based on 472 parameter rules, and (3) the generation of the speech waveform using a Klatt-type formant synthesizer. The first two processes are written in C language and implemented by a microprocessor (M 68000) and the last one is implemented by a digital signal processor (TI TMS32010). Both male and female voices can be synthesized with three different accent levels at seven different speech rates in real time. Nine kinds of subjective evaluation, which include tests for intelligibility, naturalness, and other nonlinguistic factors, were proposed and applied to the speech generated with the above-mentioned speech synthesizer. According to the results, 87.8% of the morae of the male voice and 81.6% of the morae of the female voice were identified correctly by three male subjects and two female subjects.

J19. Rhythm control based on CV-syllable positioning for Japanese synthetic speech. Toshimitsu Minowa (Corporate Engineering Division, Matsushita Communication Industry Company, Ltd., 600 Saedo-cho, Midori-ku, Yokohama, 226 Japan)

A technique has been developed for Japanese speech synthesis-by-rule to control the rhythm of synthetic speech sounds to which little attention has been given so far. In Japanese speech sounds, syllables are generally believed to be the basic elements of the rhythm, with each syllable sound pronounced almost isochronously. It was found through listening tests that there is an important portion in a syllable for recognizing the syllable and the positioning of that portion determines the rhythm. The portion was termed auditory perceptual timing point (APTP) and was determined for each syllable in listening tests. Most APTPs were found near the voice onset, which closely agreed with the result obtained by Sato [H. Sato, Trans. Comm. Speech Res., ASJ, S77-31, 1–8 (1977)]. The rhythm pattern was, in principle, determined by the number of morae in individual words and the syntactic structure of an input text, though further investigation is necessary to construct detailed rules. It has been confirmed that the quality of synthetic speech sound can be improved by employing this rhythm-control technique.

J20. Quantitative evaluation of the perceptual significance of control parameters in synthesis by rule. Yoichi Yamashita, Riichiro Mizoguchi, and Osamu Kakusho (ISIR, Osaka University, 8-1 Mihogaoka, Ibaraki, 567 Japan)

In synthesis by rule, control parameters generated by rules do not always match desired ideal values (natural sounds). To effectively improve synthetic sound quality, it is important to evaluate the perceptual significance of individual parameters and to search for the parameters most important to speech quality. This paper describes a method for measuring relative weights among some control parameters to quantitatively evaluate their perceptual significance. Perceptual weights are measured by equalizing two kinds of distances between stimuli employed in listening tests. One is defined on the physical space delimited by the synthesis parameters of stimuli. Another is defined on the psychological space delimited by parameters obtained from multidimensional scaling (MDS) techniques. MDS distributes stimuli into the space of an arbitrary dimension based on preference lists for stimuli that are obtained through listening tests. To verify the validity of the proposed method, perceptual weights for the first and second formants of the isolated vowel /a/ were measured. [Work partly supported by a Grant-in-Aid for Scientific Research on Priority Areas from the Ministry of Education.]

J21. On the unit selection measure of speech synthesis by rule using multiple synthesis units. Katsuo Abe and Yoshinori Sagisaka (ATR Interpreting Telephony Research Laboratories, Twin 21 MID Tower, 2-1-61 Shiromi, Higashi-ku, Osaka, 540 Japan)

In speech synthesis by rule, a synthesis scheme using nonuniform speech synthesis units to obtain the optimum synthesis unit sequence for a desired output speech has been proposed. In this paper, vowel spectrum variations are analyzed using the LPC-cepstrum distance to introduce a quantitative measure for unit selection. Using 5240 words, vowel spectral distortion resulting from contextual differences was studied and the following tendencies were found. (1) The neighboring phoneme, the position in the utterance, and the accentuation affect vowel spectral envelopes in this order. (2) For CVs whose following consonants have the same point or manner of articulation, the spectral distance among the vowels of

the CVs is 12% smaller than the average. (3) The vowel spectrum of the word's final CV differs from that of the word's medial CV. Based on these results, a quantitative measure is introduced to represent the spectral similarities of each vowel. With this measure, the unit selection scheme was tested using open data. Through these experiments, it was not only confirmed that the previously proposed categorical measures are adequate for general unit selection, but it was also shown that some phoneme combinations should be specially scored for unit selection.

J22. Acceptability of text-to-speech systems: Its quantification using magnitude estimation and its relationship to intelligibility and naturalness in various degrees of distortion. Chaslav Pavlovic,[a] Mario Rossi, and Robert Espesser (CNRS, Institut de Phonetique, Universite de Provence, 13621 Aix en Provence, France)

Because there are no physical measurements that quantify perceptual attributes of synthesized speech, subjective tests must be used. In this project the possibility of directly scaling acceptability via magnitude estimation is assessed. One hundred and twenty subjects took part in the study. Seven different synthesizers and three types of background noise were employed. The results are discussed in light of the following questions. (1) What is the relationship between acceptability on one hand and naturalness and intelligibility on the other in various degrees of distortion? (2) Are the objective intelligibility scores and subjective magnitude estimations highly correlated in all conditions? (3) Do the relative ratings produced by different groups of subjects agree? (4) Are the ratings on an absolute scale? (5) Are the ratings invariant to the stimulus set and range size? (6) Are the ratings practice invariant? (7) Do the ratings depend on the subject's familiarity with the test material? [a] Also at Department Speech Pathology and Audiology, University of Iowa, Iowa City, IA 52242.

J23. Quality assessment of synthetic speech using word intelligibility scores. Toshiro Watanabe (Human Interface Laboratories, NTT, Musashino, 180 Japan)

The quality of Japanese speech synthesized by rules cannot be sufficiently evaluated using only the 100-monosyllable list that has been used for speech articulation tests: Word intelligibility must also be assessed. Word intelligibility tests on both natural and synthetic speech were conducted to clarify the differences between synthetic and natural speech in learning effects, the effects of external noise, and word familiarity. Both trained and untrained listeners were used. Each word list of 155 words, selected by our previously developed method, reflected the most important word attributes of Japanese: word length, word familiarity, and initial phoneme occurrence. The experimental results show that the intelligibility scores for synthetic speech with untrained subjects were fairly low and independent of noise level. These were greatly improved by training, depending on the noise level. However, natural speech intelligibility depends little on training. The results also show that the familiarity of the test words and the first phoneme in the word are important factors in recognizing Japanese words in both synthetic and natural speech.

J24. Morphophonological derivation of Japanese predicate phrases from a semantic base. Shigeru Sato (Tohoku Institute of Technology, 35-1 Yagiyama-kasumicho, Sendai, 982 Japan)

The definition of lexical base forms may greatly affect the structure of the phonological component of a system of speech synthesis from semantic representations. This paper presents a model of mora-generating/preserving phonology for Japanese constructed from an observation of the syntactic/morphophonological derivations of predicate phrases. The achievements of the model are the following. (1) The output of the mora-independent lexico-syntax of the predicate phrase is taken care of by the three-tier rule system of word formation, phonology, and phonetics. (2) The reduction of the number of rules and the simplification of their forms are attained by transferring to syntax processes previously regarded as phonological. (3) The mora preservation phenomena are exclusively handled by the seven cyclic phonological rules. (4) Along with the rule editor system, this phonological component is fully implemented in the computer as a subsystem attached to the speech synthesis system from the semantic base [S. Sato and H. Kasuya, European Conference of Speech Technology, Vol. 2, 414–417 (1987)].

TUESDAY MORNING, 15 NOVEMBER 1988 KOHALA/KONA ROOM, 10:25 TO 11:57 A.M.

Session K. Bioresponse to Vibration II: Physiological Response to Vibration

Yasuo Tokita, Cochairman
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John C. Guignard, Cochairman
Guignard Biodynamic Associates
824 Kent Avenue
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Invited Papers

10:25

K1. Adverse effects of low-frequency ship motion: Problems of defining physiological criteria for crew protection. D. J. Thomas (Medical Director, Hoechst-Celanese, Route 202-206 North, Bedminster One Building, Somerville, NJ 08876), J. C. Guignard (Guignard Biodynamics Associates, 824 Kent Avenue, Metairie, LA 70001-4332), and G. C. Willems (Naval Biodynamics Laboratory, Box 29407, New Orleans, LA 70189-0407)

Large surface effect ship (SES) motion simulations and supporting studies using low-frequency (0.17–0.5 Hz) harmonic "heave" (vertical) motion indicated an urgent need to continue research to elucidate mechanisms of motion sickness and to develop predictive models of motion sickness incidence suitable for incorporation into human vibration exposure standards. Nineteen young male naval volunteers underwent severe SES motion simulations and a larger cohort of civilian volunteers underwent compound harmonic motion in a large-amplitude motion generator. Occurrence of and time to emesis were recorded, as well as head inertial and related biomedical responses, during the SES simulations. [The harmonic motion experiments have been reported previously by J. C. Guignard and M. E. McCauley, Aviat. Space Environ. Med. **53**, 554–563 (1982)]. Severe kinetosis, usually leading to abandonment of tasks, was observed in 16 of 19 subjects exposed to the simulated operational environment; and marked postrun illness and postural instability were frequent. Habituation to severe motion was not observed. Some correlation emerged between inertial head motion and susceptibility to sickness. Time to emesis varied idiosyncratically and is of uncertain value as a criterion of exposure severity. Further systematic research is recommended to establish the significant parameters of the physical stimulus to seasickness. [Work performed while the authors were at the NAMRL Detachment, now the Naval Biodynamics Laboratory, New Orleans, LA.]

10:45

K2. A comparative study on the response of brain monoamines to whole body vibration and local vibration in rats. Makoto Ariizumi (Department of Public Health, School of Medicine, Kanazawa University, 13-1 Takaramachi, Kanazawa, 920 Japan)

Rats were subjected to whole body vibration in the prone position, and the rats' hind legs were subjected to local vibration for 4 h. The changes in levels of the major brain monoamines, norepinephrine (NE), dopamine (DA), and serotonin (5-HT), were measured in the whole brain and in seven brain regions. Regarding the affected degree of the NE level in the whole brain, NE was decreased significantly to 57% of that of the control level by whole body vibration (20 Hz, 50 m/s^2), and was decreased only to 79% of that of the control level by local vibration (120 Hz, 50 m/s^2). DA was unaffected either by whole body vibration or by local vibration. 5-HT was remarkably elevated by whole body vibration and slightly elevated by local vibration. These changes in NE and 5-HT were observed commonly in the hypothalamus and the hippocampus. The degree of response of NE and 5-HT was much more severe in whole body vibration exposure than in local vibration exposure.

11:05

K3. Very-low-frequency vibration in a motion-base helicopter simulator compared to the simulated platform. G. O. Allgood (Martin Marietta Energy Systems, Inc., Oak Ridge, TN 37830), R. S. Kennedy, and K. S. Berbaum (Essex Corporation, 1040 Woodcock Road, Orlando, FL 32803)

The physiological symptoms of simulator sickness parallel those of true motion sickness, and are being reported with increasing regularity in Navy flight trainers, particularly helicopter simulators. It was hypothesized that nauseogenic energies in the very-low-frequency (VLF) domain (viz., < 1 Hz) would be present in moving base helicopter simulators but not in the actual aircraft that should have energies present at higher frequencies (viz., > 2 Hz). A Navy helicopter and its simulator were instrumented to record frequency of acceleration and acceleration profiles with pilots-in-the-loop. Motion sickness was reported in the simulator where recorded energies exceeded limits for VLF vibration directed in Military Standard 1472C (MIL-STD-1472C). No sickness was recorded for pilots flying the helicopter where vibration profiles were at higher frequencies. The presence of VLF vibration in flight simulators and not in aircraft is implied as a cause of simulator sickness. Development of an advanced biomechanical motion analyzer could be employed to signal when on-line devices exceed human exposure limits as well as for test and evaluation. Furthermore, such data could inform future revisions of military standards for dynamic vehicle operations.

11:25

K4. Human physiological response to low sound-pressure level infra- and low-frequency noise. Yasuo Tokita[a] (Kobayasi Institute of Physical Research, Kokubunji, 185 Japan)

During the last 20 years or so, there have been many complaints about infra- and low-frequency noise as an environmental problem in the neighborhood of factories, long road bridges, civil engineering blast sites, high-speed train tunnels, etc. The correlation between the physiological response in laboratory experiment results and field surveys on complaints is important in evaluating the infra- and low-frequency noise. This study has been with pure tone and band noise exposure (frequency: 2–500 Hz; level: 50–110 dB SPL) on the following items: (1) respiration and pulse, (2) blood pressure, (3) evoked potentials of brain waves, (4) stress hormones, and (5) sleep interference. According to these results, the quantitative relation between the human responses and sound-pressure level and frequency are not clear but, among these items, sleep interference has a rather better correlation with the sound-pressure level and frequency than the other items, and corresponds with the results of psychological experiments. [Work supported by Environment Agency.] [a] Present address: Aircraft Nuisance Research Center, Haneda-kuko, Ohta-ku, Tokyo, 144 Japan.

11:45

K5. Shipboard rest, sleep, and seasickness: Evidence of interaction. A. C. Bittner, Jr. (Analytics, Inc., 2500 Maryland Road, Willow Grove, PA 19090) and J. C. Guignard (Guignard Biodynamics Associates, 824 Kent Avenue, Metairie, LA 70001-4332)

Sea-keeping trials of a United States Coast Guard cutter included a human factors engineering evaluation having particular regard to problems of seasickness in certain work stations. Some motion sickness incidence and related data, and the derivation of human factors engineering principles therefrom, were reported in 1985 [A. C. Bittner, Jr. and J. C. Guignard, Nav. Eng. J. 97(4), 207–213; 97 (5), 107–111]. Further analysis [A. C. Bittner, Jr. and J. C. Guignard, in *Trends in Ergonomics/Human Factors V*, edited by F. Aghazadeh (Elsevier, New York, 1988), pp. 529–539] showed that, contrary to traditional assumptions, seasickness is characterized by at least two functionally independent factors, identified as $F1$ (symptomatic general motion illness) and $F2$ (retching-vomiting). It was shown that the validity of future sea trials requires: (a) multiple-score scaling of motion sickness (e.g., using $F1$ and $F2$); (b) control of subject crew activity and movements about the ship during periods of observation; and (c) avoidance of steaming patterns that induce extraneous carryover effects. The present paper draws attention to an incidental finding concerning the influence of brief sleep and rest (lying down in the bunk) upon motion sickness susceptibility and recovery in the shipboard environment. Analyses of questionnaire data including amount and subjective quality of sleep showed that these were affected by the passage of time in an adverse (nauseogenic) motion environment, as has been observed generally in naval operations. An apparent ameliorative effect of unscheduled rest periods and naps upon susceptibility to adverse motion effects was incidentally detected and is commented upon in this report. [Observations made while the authors were at the Naval Biodynamics Laboratory, New Orleans.]

Session L. Engineering Acoustics II: General Acoustical Measurement Techniques

Robert D. Finch, Cochairman
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Contributed Papers

2:00

L1. Experimental evaluation of dynamic fatigue in glass reinforced composite flextensional transducer shells. Kim Benjamin and James Sturges (Raytheon Company, Portsmouth, RI 02871-1087)

Results from a controlled experiment aimed at determining the fatigue life, critical flaw size, and critical flaw location for glass reinforced composite flextensional shells are presented. Four shells, two E-Glass and two S-Glass were built into active projectors and cycled in air for 10^9 cycles. Intentional interlaminar and surface type flaws were symmetrically located on a shell of each composition. Static and dynamic stress levels simulated the anticipated conditions in water at full dynamic displacement and maximum operational depth. The fatigue failure criterion was defined as a 15% change in shell stiffness. The stiffness in the shells was computed from the measured transducer admittance and strain gage readings. Other sensory data included those from accelerometers, thermistors, and pre- and postcycling color photographs of the shell. The results indicate that all four shells survived one billion cycles without any significant performance degradation. In addition to these fatigue data, creep measurements obtained on the shells and ceramic stacks are also presented.

2:12

L2. Evaluation of the complex coefficients $\hat{s}_{33}^H$, $\hat{d}_{33}$, and $\hat{\mu}_{33}^T$ of length expander magnetostrictive rods. Donald Ricketts and Kim Benjamin (Raytheon Company, Portsmouth, RI 02871-1087)

An analytical model is presented for evaluating the effective complex material coefficients of length expander piezomagnetic rods with magnetic field in the direction of wave propagation. In particular, the model facilitates the evaluation of complex $\hat{s}_{33}^H$, $\hat{k}_{33}$, $\hat{\mu}_{33}$, and $\hat{d}_{33}$ from the measured set of quantities $[f_R^I, Q_M^I, f_R^E, Q_M^E, Z_{ex}, f_{ex}]$, where the superscripts I and E denote the ac open-circuit and short-circuit conditions, respectively. Thus f_R^I and Q_M^I are the measured ac open-circuit resonance frequency and mechanical storage factor; likewise for f_R^E and Q_M^E. Here, Z_{ex} is the measured value of the electrical impedance at the frequency f_{ex}. An impact testing technique [J. Acoust. Soc. Am. Suppl. 1 **67**, S32 (1980)] is used to measure the resonance frequencies and mechanical storage factors of the free–free magnetostrictive rod test specimen for the two electric boundary conditions. Experimental results are presented for $\hat{s}_{33}^H$, $\hat{d}_{33}$, and $\hat{\mu}_{33}^T$ of grain-oriented Terfenol-D rods for several levels of magnetic bias.

2:24

L3. Reciprocity calibration of a Tonpilz transducer by the Delta-Z method. Mark D. Patton,[a)] Steven R. Baker, and Oscar B. Wilson (Department of Physics, Code 61Ba, Naval Postgraduate School, Monterey, CA 93943)

The Delta-Z reciprocity calibration method [R. Bedard and S. R. Baker, J. Acoust. Soc. Am. Suppl. 1 **83**, S19 (1988)] has been applied to a Tonpilz transducer. The diffraction constant and the free-field radiation impedance in water were calculated using the U.S. Navy's CHIEF computer code [H. A. Schenck, J. Acoust. Soc. Am. **44**, 41 (1968)]. Measurements of the input electrical impedance in water and in air, and a standard comparison calibration were performed at the Transducer Evaluation Center (TRANSDEC), Naval Ocean Systems Center, San Diego, CA. The results of the Delta-Z method and the standard comparison calibration method will be compared. [Work sponsored by NRL-USRD.] [a)] LT, USN.

2:36

L4. Measurement of flow resistance of porous materials at high flow rates. Anthony G. Galaitsis (Bolt Beranek and Newman, 10 Moulton Street, Cambridge, MA 02238) and Mark Kiss (Naval Underwater Systems Center, Newport, RI 02840)

A flow resistance measurement facility was developed to quantify the flow characteristics of porous sheets under pressure differences of up to 100 psi. These materials have been utilized in a particular class of silencers developed to reduce the noise of impulsive, high-pressure air discharges. The silencers consist of a porous-wall enclosure of volume V, they yield an insertion loss IL, and develop a backpressure ΔP. The porous material properties near the operating pressure must be known accurately to minimize (through analytical design) the silencer volume while meeting insertion loss and backpressure requirements. Prior to developing this flow facility, the analytical design was performed by extrapolating the material properties measured by manufacturers at low pressures (about 0.08 psi), which lead to uncertainties and, often, required design iterations. The usage of the flow facility allows accurate determination of material properties near the operating point and improves the efficiency of the design process. The presentation will describe the flow facility and recent test results, and will review the silencer design procedure.

2:48

L5. Acoustical properties of drill strings: Theory. Douglas S. Drumheller (Geothermal Research Division 6252, Sandia National Laboratories, Albuquerque, NM 87185)

Drill strings are assembled from 30-ft sections of steel pipe and used to drill wells to recover petrochemical and geothermal resources. The ability to transmit data by acoustic carrier waves generated within the drill string would have significant economic impact on the drilling industry. This poses an unusual waveguide problem. The string, which may be up to 30 000 ft long, forms a periodic structure of slender pipe threaded together with large-diameter tool joints. Classical analysis methods along with numerical algorithms can be used to study the behavior of both periodic and transient acoustic waves in this structure. Stress waves that propagate along this waveguide exhibit all of the phenomena associated with Brillouin scattering theory. The results show a comb filter structure with passbands that produce significant wave dispersion. In addition, the acoustic impedance that determines the design of sending and receiving transducers is a strong function of position as well as frequency. [This work was supported by the U.S. Department of Energy at Sandia National Laboratories under Contract DE-AC04-76DP00789.]

3:00

L6. Acoustical properties of drill strings: Application. Douglas S. Drumheller (Geothermal Research Division 6252 Sandia National Laboratories, Albuquerque, NM 87185)

Analysis has demonstrated a viable data transmission system for communicating by acoustical carrier waves in drill strings used to drill wells. Because of the periodic structure of this waveguide, complex transmission characteristics exist. Two experiments have been used to study these phenomena. In the first experiment, a scale laboratory model was constructed to examine the acoustical response of this system. A longitudinal impulse was produced with a hammer and the resulting signal was measured at several locations with semiconductor strain gauges. In the second experiment, the impulse response of a 1500-ft drill string suspended in a mud-filled well was tested. The data exhibit all of the phenomena predicted by the theory. Measured attenuation is about 2 dB per thousand feet. Both the comb filter phenomena produced by the periodic structure of the pipe and tool joints, and the standing-wave interference pattern caused by echos between opposite ends of the drill string are present. [This work was supported by the U. S. Department of Energy at Sandia National Laboratories under Contract DE-AC04-76DP00789.]

3:12

L7. Noise measurements using a portable instrument controller. Frank H. Brittain and Mark M. Gmerek (Bechtel National Inc., P.O. Box 3965, San Francisco, CA 94119)

Sound-level meters are very useful and flexible for brief measurements in the hands of an experienced operator, but measurements for periods of an hour to several days are often not practical. While community noise monitors are useful for measurements of longer duration, they typically lack the flexibility for many specialized measurements. The desired flexibility is obtained by interfacing a sound-level meter to a portable computer. The two are connected through an RS-232 interface. The computer is programmed to control and collect data from the sound-level meter. The data are stored and then formatted and printed or transferred to a larger computer. The need to write data by hand is eliminated. Several examples are given to illustrate the technique.

3:24

L8. Compensation for phase mismatch in sound intensity measurements. Gordon Ebbitt (Bruel & Kjaer, DK-2850 Naerum, Denmark)

Sound intensity measurements place stringent requirements on the phase matching of the measuring apparatus (including the transducers). In the past this has nearly always been accomplished by relying on the manufacturer to provide phase matched instrumentation and transducers. These days more and more processing power is being built into analyzers. This has opened the door to the possibility of compensating for the phase mismatch between the two measurement channels. The procedure consists of first measuring the phase mismatch between the channels and then correcting for it as a postprocessing operation. In practice, this correction can be included as part of the calculations required to compute the intensity from the two input signals. While there are clear benefits to this technique (matched transducers are not required), there are a number of practical problems. Chief among these is getting an accurate measurement of the phase difference between the channels. The benefits and difficulties associated with this technique will be discussed and measurements made with a phase-matched system will be compared with those made using phase compensation.

3:36

L9. An application of the Wigner distribution to loudspeaker evaluation. Kazuhiro Nakamura and Takeshi Miyagawa (Corporate Engineering Division, Matsushita Communication Industry Company, Ltd., 600 Saedo-cho, Midori-ku, Yokohama, 226 Japan)

The problem in the sound quality evaluation of loudspeakers using their frequency characteristics is that the sound quality difference perceived by human ears is not distinctly represented in the frequency characteristics. To solve this problem, an evaluation method using the Wigner distribution that is derived from the impulse responses of loudspeakers is employed. The effectiveness of this evaluation method in comparison with conventional frequency characteristics was examined using two loudspeakers with clearly different sound qualities. While there was no noticeable relationship between the frequency characteristics of the loudspeakers and listener's subjective impressions, the Wigner distributions showed a recognizable correlation with listener's subjective impressions. Thus it can be concluded that the Wigner distribution can represent the sound qualities of loudspeakers more effectively than conventional frequency characteristics.

3:48

L10. A study on a detection method for chemically solidified soil using impulsive sound. Tsutomu Watanabe, Takuya Takahashi (Department of Civil Engineering, Chiba Institute of Technology, 2-17-1 Tsudanuma, Narashino, 275 Japan), and Seiichi Motooka (Department of Electronics, Chiba Institute of Technology, 2-17-1 Tsudanuma, Narashino, 275 Japan)

In engineering foundation works, the chemical grouting method is useful to increase the bearing capacity of the soil. If the shapes and the location of the underground solidification caused by chemical grouting can be detected without digging up the ground, it will result in greater safety in engineering work operations. The authors devised a method to detect the location and the shapes of solidification by applying a pulse-echo method using impulsive sound. In this method, a hole (diam, 100 mm; depth 3 m) is dug vertically about 1 m from the location of the chemical grouting. The hole is filled with water and an electromagnetic induction-type sound source is suspended in it. The echo signals from the chemical solidification are received by four hydrophones suspended symmetrically with respect to the sound source. These outputs are processed by a technique similar to polarity correlation. Then, two-dimensional image patterns of the solidification can be obtained. Experiments in a model sand bath ($300 \times 300 \times 300$ cm) show satisfactory results.

4:00–4:10

Break

4:10

L11. Design of an automated, computer-controlled acoustic tank facility and its application to measurement of nonlinear effects. Gary P. Schwaiger, Jerry H. Ginsberg, and Peter H. Rogers (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

A computer-actuated acoustic research tank facility designed for facilitating precise acoustic measurement of sonar transducers, small arrays, and structural models has been installed at the Georgia Institute of Technology. The apparatus consists of a wooden tank, having dimensions of 7-m diameter and 5-m depth, integrated to a computer actuated positioning and data retrieval system. A resident minicomputer controls positioning and signal generation, as well as providing temporary data storage and signal processing capabilities. Permanent data storage and post-processing are provided by a remote minicomputer. The multiaxial positioning and scanning system, which extends over the entire active tank area, provides accurate triaxial and rotational positioning of the transducer. Component hardware, positioning system design features, computer software and facility capabilities are presented. One current use for the facility is to perform a series of experimental measurements exploring nonlinear effects encountered in sound beams generated by a high-power transducer operating at a moderately high *ka* value. Waveforms are measured at a variety of locations in the near and farfields, both on and off axis. Signal processing then provides amplitude and phase information for the fundamental and higher harmonics.

4:22

L12. Performance of a continuous transmission frequency modulation (CTFM) in-air ultrasonic system applied to dynamic production processes. Robert D. Collier, Roderick C. MacNeil (Department of Mechanical Engineering, Tufts University, Medford, MA 02155), and Leslie Kay (Consultant, Sensonix Inc., 70 Walnut Street, Wellesley, MA 02181)

The measurement of the thickness of a continuous, moving sheet of deformable material on rollers and conveyor belts is required to control sheet thickness. Laboratory measurements demonstrate accuracies of ± 0.005 in. with articulated separate transmitter and receiver transducers at distances of 2–7 in. with a CTFM signal 200–100 kHz. System performance as a function of temperature, alignment, distance, and target reflection characteristics are described. Laboratory dynamic simulation of production processes demonstrate capabilities to measure both step and continuously variable material profiles for conveyor speeds in the range of 30 fpm. Results are compared with *in situ* production measurements.

4:34

L13. Development and test of piezoelectric vibration calibrators to be used for frequency range above 10 kHz. Li-Feng Ge (Anhui Bureau of Standards and Metrology, Hefei, Anhui Province, People's Republic of China)

Piezoelectric calibrators intended as vibration standards at frequencies higher than 10 kHz, based on the reciprocity method, have been developed and tested. Each calibrator is composed of a table, a two-disk active element, and a base. The first resonance frequency of the shakers is higher than 30 kHz; and the capacitance of the electrical port is about 10–15 nF. The design possesses many advantages, such as: simple structure, stable electrical and mechanical characteristics, low distortion, and good amplitude linearity. In particular, the electrical impedance changes resulting from mass load changes are sufficiently large and repeatable in the higher frequency range, 10–40 kHz, to permit accurate calibration by the piezoelectric reciprocity method.

4:46

L14. Design of an ultrasonic wattmeter. Seiji Kaneko, Masahide Gakumazawa (Department of Electrical Communication, Shibaura Institute of Technology, 3-9-14 Shibaura, Minato-ku, Tokyo, 180 Japan), and Kiyoshi Shinoda (Towa Electric Co., Ltd., Saitama, 352 Japan)

A new type of ultrasonic wattmeter, which employs a multiplier IC, has been designed based on the fact that the power is represented by a product of the voltage, the current, and the phase in a high-frequency electric circuit. The output of the IC is averaged in time by a low-pass filter to give a dc signal proportional to the power. The present wattmeter, unlike the conventional ones, has the advantage that the measurements are made over a frequency range from 1 kHz to 1 MHz and also at a high voltage load. This paper describes the outline of the wattmeter and its characteristics.

4:58

L15. The effect of tissue attenuation on Doppler flow imaging with linear phased array. Xuguang Lee and Shusen Ji (Department of Precision Instruments, Shanghai Jiao Tong University, Shanghai, People's Republic of China)

In Doppler flow imaging with linear phased array, each scan line is implemented on the basis of the pulsed Doppler principle. Transmitted pulses are backscattered by blood flow. Owing to a relatively deeper depth of detection, the backscattered Doppler signal must propagate a longer distance in soft tissues, which causes the acoustic signal to be influenced by the tissue's frequency-dependent attenuation characteristics. Consequently, a distortion of the power spectra of the backscattered Doppler signal appears. Unfortunately, there are little or no definitive data that address this problem. With the model of nonlinear frequency-dependent attenuation, the erroneous values of the image are predicted and an error matrix is presented to consider the scan conversion process. It is concluded that the mean frequency error of a sample volume depends on not only the attenuation characteristics along the propagation path but also on the flow direction relative to the probe. In the forward flow case, the mean frequency is shifted toward lower frequencies, while reverse flow causes the mean frequency to be shifted toward higher frequencies.

5:10

L16. Phase technique for calibration of large, low-frequency underwater sound projectors. G. D. Hugus (Naval Research Laboratory, Underwater Sound Reference Detachment, P.O. Box 568337, Orlando, FL 32856-8337) and D. L. Carson (Texas Research Institute, Inc., 2761 Evergreen Street, San Diego, CA 92106)

The acoustic source level of high-power, low-frequency (10–250 Hz) underwater sound projectors is difficult to measure accurately. This is primarily because of the inaccuracies in measurement of the distances between the effective acoustic centers of the projector and measurement hydrophones. These distances (up to 25 m) cannot be easily determined by direct measurement because acoustic measurements on these inherently large projectors are typically made in open water calibration facilities or at sea. A technique has been developed to measure these large distances more accurately by comparing the phase of two different hydrophone outputs in the farfield. This technique depends on the assumption that the projector acts as a fixed point source and is therefore omnidirectional at these low frequencies, and that free-field conditions exist. [Work supported by Space & Naval Warfare Systems Command.]

5:22

L17. Free-field acoustic calibration of long underwater acoustic arrays in a closed tube. L. D. Luker, J. F. Zalesak, and C. K. Brown (Naval Research Laboratory, Underwater Sound Reference Detachment, P.O. Box 568337, Orlando, FL 32856-8337)

A technique is described for measuring the complex free-field sensitivity and directivity patterns of long underwater acoustic arrays in a closed tube. The tube contains sound projectors and monitoring hydrophones located along its length. Peforming these free-field calibrations normally requires a large body of water to insure that the reflections from the sides, surface, and bottom have a negligible effect on the measurement. The projectors are driven to produce an acoustic pressure in the tube that is equivalent to a free-field plane acoustic wave propagating from any angle of incidence. The required projector drives, both amplitude and phase, are determined by measuring the transfer function for every projector and every hydrophone location and inverting the resulting transfer matrix. A prototype 10-m-long system was developed. The system operates over the frequency range from 100–2000 Hz and is capable of simulating depths to 600 m. Directivity patterns, obtained using the prototype, for a test line array 5 m long were in good agreement with theoretically predicted patterns. [Work supported by ONT.]

TUESDAY AFTERNOON, 15 NOVEMBER 1988 — MAUI ROOM, 2:00 TO 4:39 P.M.

Session M. Noise I and Architectural Acoustics III: Application of the Sound Intensity Technique

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Chairman's Introduction—2:00—C. I. Holmer

Invited Papers

2:05

M1. Avoiding pitfalls in mastering sound intensity measurements. Frank H. Brittain (Bechtel National Inc., P.O. Box 3965, San Francisco, CA 94119)

Sound or acoustic intensity measurements yield results that were previously impractical or, in some cases, not feasible to obtain from measurements of sound-pressure level. Unfortunately, these techniques are also more sophisticated and present more opportunities to get erroneous results. Errors of 10 dB and more are not uncommon, nor is it rare to obtain irrelevant data. Because intensity is measured indirectly, these errors are often not obvious. To get results sufficiently close for engineering purposes, a considerable investment in instrumentation, education, and practice is required. This paper suggests how to begin the process of mastering intensity technology, including suggestions for initial reference materials. A series of experiments is suggested not only to develop proficiency, but to verify validity of the results determined. Examples are given to illustrate limitations of intensity techniques.

2:25

M2. Evaluation of measurement accuracy of a sound intensity probe. Hideo Suzuki (Ono Sokki Co., Ltd., 2-4-1 Nishi-shinjuku, Shinjuku-ku, Tokyo, 163 Japan)

The two-microphone sound intensity measurement technique is most commonly used. There are several causes of measurement error with this technique. Some of them are inherent in the technique itself, and others are due to the imperfect performance of the microphone probes. Various types of measurement errors are reviewed first. Then, the measurement accuracy of a one-dimensional, four-microphone probe is discussed. The accuracy of the intensity measurement can be evaluated either in an acoustic tube or in an anechoic chamber. The effective distances between pairs of microphones, active and reactive intensities, and particle velocities measured by use of the probe in an anechoic chamber are presented. These results, including the directivity patterns, are compared with those of a conventional intensity probe. The residual intensity index of the present probe is also presented.

2:45

M3. A study on the standard for sound intensity measurement. Takeshi Fujimori, Sojun Sato, and Hajime Miura (Electrotechnical Laboratory, Tsukuba, 305 Japan)

A study on a traceability system for sound intensity standards required for accurate sound intensity measurements is described in detail. Required characteristics for the system, such as a calibration method for sound intensity meters involving a field calibrator and a working standard, are discussed. A free-field calibration method using a progressive wave or a standing wave in an anechoic room is proposed. This is valid for the estimation of accuracies of intensity measurements in complicated actual sound fields. Some ideas on the generation of a calibrating sound field and a standard sound intensity probe are given. The standard probe is composed of several pressure microphones mounted on a rigid sphere and is precisely calibrated using a laboratory standard condenser microphone. To obtain enough accuracy in intensity measurements using the system, five sound intensity meters on the market were tested in various sound fields. The results show that the proposed traceability system is valid for sound intensity measurements.

3:05

M4. The use of intensity techniques for noise source identification in small machinery. U. D. Dietschi (Clark Laboratory Services, 821 East Front Street, Buchanan, MI 49107) and J. S. Bolton (Ray W. Herrick Laboratories, School of Mechanical Engineering, Purdue University, West Lafayette, IN 47907)

Intensity techniques are now widely used for two purposes: sound power determination and source identification. The work described in this presentation illustrates an application of the latter type. As part of a noise control program for a small domestic appliance, a narrow-band intensity measurement system was developed based on a face-to-face microphone probe, a two-channel FFT analyzer, and a personal computer. The probe was used to measure the nearfield normal intensity on planes parallel to the device's major surfaces. Graphical display of these data allowed the radiated sound field to be visualized at any desired frequency. Further, it was possible to distinguish between various types of component sources by examining the field in detail. For example, airborne sound that is generated within the device and radiates to the exterior through apertures or "leaks" was plainly visible. Radiation from the exterior surface of the device resulting from direct vibrational excitation of the surface and from structure-borne vibration could also be identified. Knowledge of this type was used to advantage to guide the noise control treatment program. Nearfield surveys conducted after these modifications yielded a particularly vivid indication of the effects and benefits of the individual modifications.

3:25

M5. Uses of sound intensity in architectural acoustics practice. Steven J. Orfield (Orfield Associates, Inc., 4551 Bloomington Avenue, Minneapolis, MN 55407)

Sound intensity has received significant attention over the past five years in the fields of noise control and product noise analysis. Yet, at this date, it is an infrequently used technique in consulting acoustics in general. Standards have been slow in developing and thus have not increased its use, and the most fundamental reason for its use is still the utility of its results rather than any existing standard. The benefit of intensity measurement in architecture is principally in its ability to indicate source direction, and this benefit is particularly important in a number of fields now being studied. The first of these is the open plan office, and the current work in intensity is based on an interest in the performance of acoustical dividers and absorbers. The second field, for which intensity seems to be the only measurement technique that is easily used, is the analysis of compound element building facades. This analysis is aimed at developing field transmission loss values for facade elements, such as windows, doors, and wall sections.

3:45

M6. Sound intensity measurement for impulsive sounds. Hideki Tachibana and Hiroo Yano (Institute of Industrial Science, University of Tokyo, 7-22-1 Roppongi, Minato-ku, Tokyo, 106 Japan)

For the measurement of sound radiation properties caused by structure-borne sound from large structures such as buildings and bridges, impact excitation is much more advantageous and convenient than stationary random excitation. The application of the sound intensity method to such sound radiation measurements was investigated. First, it was theoretically clarified that the result obtained by time integrating the sound intensity impulse response (instantaneous sound intensity) measured by impact excitation corresponds to the sound intensity obtained by stationary random excitation under the conditions of linear vibration propagation and sound radiation. This theoretical relation was examined and confirmed by a basic model experiment in which sound power and sound energy radiation patterns from a steel I beam were measured by both of the exciting methods. As a practical application, the noise radiation properties of an iron railroad bridge were investigated. In this experiment, the rail was impacted by an iron hammer at a constant strength, and the sound intensity around the structure was measured and integrated. From this measurement, the effect of vibration isolation of the rails was examined.

Contributed Papers

4:05

M7. Identification of multiple point sources using acoustic intensity. Tsuyoshi Usagawa, Hiromitsu Miyazono, and Masanao Ebata (Faculty of Engineering, Kumamoto University, 2-39-1 Kurokami, Kumamoto, 860 Japan)

The identification of noise sources, especially in a small instrument, is important for effective noise control. When the noise radiated from a small instrument is analyzed, acoustic intensity data provide more information than conventional pressure measurement because the acoustic intensity is a vector quantity of the acoustic energy flow. This paper presents a method for the identification of multiple point sources using acoustic intensity, and discusses the performance of the proposed method using a model experiment. The proposed method is based on the assumption that an observed intensity vector is given as a summation of individual intensity vectors radiated from each noise source. This method identifies multiple source positions and estimates the intensity of each source simultaneously. In the model experiment, the proposed method separates three point sources, which are 60 mm apart from each other and radiate 500-Hz sinusoidal waves.

4:17

M8. On the relationship between sound intensity, pressure, particle velocity, and measurement accuracy. J. Pope (Bruel & Kjaer Instruments, Inc., 185 Forest Street, Marlborough, MA 01752)

Sound intensity measurements are sensitive to instrumentation inaccuracies in a manner that varies with the sound field in which the meaurements are made. For this reason a number of "quality indicators" have been proposed to allow evaluation of the uncertainty of a measured intensity, the most widely used indicator being the "pressure-intensity index." Intensity is usually calculated from measurements of sound pressure and either pressure gradient (i.e., two-microphone, p-p devices) or particle velocity (p-u devices). The interpretation and use of the pressure-intensity index is explored for such devices, with emphasis on practical use. It is shown that this indicator is a "phase index" for p-p devices, and an "impedance index" for p-u devices. It is shown also that proper interpretation of the indicator varies with the manner in which the measurement device is realized. For example, p-p devices that sum and difference before filtering require interpretation different from those that filter first. It is concluded that universal limits to ensure a given measurement accuracy cannot be established independent of the measuring device.

4:29–4:39

Chairman's Summary—H. Tachibana

Session N. Physical Acoustics II: Nonlinear Acoustics, Part II

Lawrence A. Crum, Cochairman
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The Graduate School at Nagatsuta
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Contributed Papers

2:00

N1. Dispersion and attenuation of wind-driven deep gravity waves. Andres Larraza (Department of Physics, Naval Postgraduate School, Monterey, CA 93943) and Seth Putterman (Department of Physics, University of California, Los Angeles, CA 90024)

Field and laboratory measurements of the surface of the ocean driven by the wind yield phase velocities that differ from the linear theory dispersion law by as much as 60% [M. Ramamonjiarisoa and C. A. Coantic, Acad. Sci. Paris **288**, 111 (1976) and V. Yefimov *et al.*, Atmos. Ocean. Phys. **8**, 435 (1972)]. Both short gravity waves and long low-frequency waves are generally observed. The modulations induced by the long wave may have substantial influence on the space-time structure of the high-frequency noise. The wave turbulence motion then becomes inhomogeneous and anisotropic. The short waves in turn modify the surrounding medium by virtue of their radiation stress. The nonlinear interaction between these two components of the sea wave spectrum leads to two propagating modes both of which have characteristics that differ substantially from the linear theory dispersion law. Comparison of the theory with experimental results will be made; further experiments will be suggested.

2:12

N2. Observation of a topological kink soliton in a highly compliant waveguide. Bruce Denardo, Andrés Larraza, and Seth Putterman (Department of Physics, University of California, Los Angeles, CA 90024)

A localized nonpropagating 180° kink has been observed in the phase of the (0,1) cross mode of a parametrically driven trough of liquid. The depth must be less than approximately the width of the trough, so that the resonance hardens. Nonpropagating breather solitons have been observed in a similar geometry but in the deep liquid limit, where the resonance softens [Wu *et al.*, Phys. Rev. Lett. **52**, 1421 (1984)]. A kink is insensitive to the ends of the trough unless it is near an end, where it is repelled. If the amplitude is continually decreased, the node is forced to the center, and the wave becomes the linear (1,1) mode. A kink and antikink in close proximity repel each other. In a trough that has a slowly varying width or depth, a kink slowly migrates toward the region of smaller cutoff frequency. Theoretically, modulations of the cross mode are described to lowest order by the nonlinear Schrödinger equation with the opposite sign of the nonlinear term. This possesses a "dark" soliton solution [Hasagawa and Tappert, Appl. Phys. Lett. **23**, 171 (1973)] of which the kink is a special case. [Work supported by DOE.]

2:24

N3. Recent progress on nonpropagation hydrodynamic solitons. Ronjue Wei, Benren Wang, Yi Mao, Xiaoyu Zheng, and Guoqing Miao (Institute of Acoustics, Nanjing University, Nanjing, People's Republic of China)

The present report will emphasize the recent development of the generation of nonpropagating hydrodynamic solitons of Faraday's oscillating water tank experiment first observed by Wu *et al.* [Phys. Rev. Lett. **52**, 1421 (1984)] in a small rectangular trough ($38\times2.7\times2$ cm^3) driven vertically and simple harmonically with a vibrator of fundamental frequency of 10 Hz. It will include: (1) the effect of the geometrical dimension, driving frequency, and amplitude on the stability region, etc. of the solitons, (2) further study on the multisoliton collisions, and (3) the condition and behavior for this solitary system that exhibits bifurcations and chaos, which therefore provide experimental evidence of the relation between these two dominant nonlinear phenomena. Some theoretical models are proposed as well as questions raised by this experimental simplicity. [Work supported by Chinese National Science Foundation.]

2:36

N4. Simulated focusing/beamforming with a phased array of nonlinear sources. B. Edward McDonald (NORDA, NSTL, MS 39529) and W. A. Kuperman (NRL, Washington, DC 20375)

Numerical simulations have been carried out on ocean acoustic scales for a simple array of three nonlinear sources. The nonlinear progressive wave equation (NPE) model [B. E. McDonald and W. A. Kuperman, J. Acoust. Soc. Am. **81**, 1406–1417 (1987)] has been used to calculate the evolution of initially nonintersecting underwater shocks of Mach number 0.05. Three impulsive sources equally spaced on a vertical line are fired with top and bottom elements together, and the middle one phase delayed. The resulting wave produces a weak "focus" at the intersection of the three expanding spherical waves. Linear and nonlinear calculations are compared. In the linear calculation, the individual waves retain their identity past the focal range. In the nonlinear calculation, however, Mach stems form as individual shocks merge. A directed beam is formed, with the three shocks joined irreversibly into a single shock. Nonlinear processes are seen to increase peak pressures in the farfield as compared to a linear solution.

2:48

N5. Wave amplification in chemically active bubble media. F. N. Zamarayev, V. K. Kedrinskii (Lavrentyev Institute of Hydrodynamics, Novosibirsk 630090, USSR), and Ch. L. Mader (Los Alamos National Laboratory, Los Alamos, NM 87545)

A numerical simulation is performed for shock propagation in chemically active two-phase "liquid-explosive gas bubbles" media. It is established experimentally that this process is accompanied by formation of a stationary disturbance of the wavetrain form. It propagates with constant velocity exceeding essentially the equilibrium velocity characteristic of the main disturbance. Its maximum amplitude is practically constant and exceeds significantly the amplitude of an incident wave. Such a wave is called a bubble detonation wave or secondary detonation. The basis of this effect is the mechanism of separation of the shock wave in a bubble medium into the main disturbance and precursor [V. K. Kedrinskii, PMTF **4**, 29–34 (1968)]. The latter decays quickly. However, in chemically active media these losses are compensated for by the bubble explosion energy, thereby providing the existence of a self-sustaining regime. The present study is concerned with two approaches. The first one considers the interaction of a strong plane shock wave with a single spherical bubble filled in with a one-to-one molar acetylene–oxygen mixture. It is shown that the refracted wave can initiate this mixture's detonation [V. K. Kedrinskii and Ch. Mader, Proc. XVI Int. Symp. on Shock Tubes and Waves, July 1987, Aachen, Federal Republic of Germany]. The mixture ignition due to adiabatic bubble compression by a weak shock wave is analyzed within the framework of relatively simple kinetics. The second approach is based on a two-phase model and considers the process of formation of the stationary disturbance in a semi-infinite active bubble medium. The results obtained are in good agreement with the experimental data [A. I. Sychev and A. V. Pinayev, FGV **3**, 22 (1986)].

3:00

N6. Transfer matrix theory of the propagation of leaky waves in periodic or inhomogeneous media. Didier Sornette, Louis Macon, and Jean Coste (Laboratoire de Physique de la Matière condensée, CNRS UA 190, Faculté des Sciences, Parc Valrose, 06034 Nice Cedex, France)

A general transfer matrix approach for the propagation of guided waves in the presence of inhomogeneities or "scatterers" is presented that particularly addresses the problem of coupling with radiation modes leading to a leakage of the guided wave in the bulk at each scattering. From symmetry and conservation laws, the general form of the transfer matrix is obtained in terms of four independent real parameters. For one-dimensional periodic lattices of identical scatterers, the leakage vanishes at the band edges: This *coherent effect* stems from the complete destructive interference between the radiations of the different scatterers at these particular points. The competition between Anderson localization and coherent leakage in the presence of disorder is discussed. Experimental results on the propagation of one-dimensional surface acoustic waves in quasiperiodic and random systems of parallel grooves are presented. Spectral and temporal properties of localization in such systems are well put in evidence in this model experimental system. [Work supported by DRET.]

3:12

N7. Attempts to observe acoustic wave turbulence. Bruce Denardo, Francois Gallet, Seth Putterman (Department of Physics, University of California, Los Angeles, CA 90024), and Albert Migliori (Los Alamos National Laboratory, Los Alamos, NM 87545)

High-amplitude, low-frequency, broadband acoustic excitation of a medium should lead to a cascade of energy that is analogous to the Kolmogorov turbulence that characterizes vortex eddies. The acoustic cascade should also lead to the phenomenon of classical second sound [Larraza and Putterman, Phys. Rev. Lett. **57**, 2810 (1986)]. Attempts to observe the turbulent acoustic power spectrum $e(\omega) \propto \omega^{-3/2}$ and higher-order correlations, in controlled laboratory experiments, are being carried out in the Leo P. Delsasso Acoustics Labs at UCLA. The Allen and Rudnick high-frequency siren [J. Acoust. Soc. Am. **19**, 857 (1947)] is being used as a source. In the reverberation room amplitudes of about 160 dB have been achieved. While effects of higher harmonics and scattering of sound by sound can be seen, the intensity level is still apparently about 10 dB below the threshold for acoustic turbulence. [Work supported by DOE.]

3:24

N8. Multidimensional large deformation waves by the theory of characteristics. M. Ziv (Holon Technological Institute, 52 Golomb Street, P. O. Box 305, Holon 58368, Israel)

A presentation is given of solutions to multidimensional wave propagation in solids. These solutions are obtained by a method based on the theory of characteristics. While for one-dimensional analysis a characteristics formulation suffices, it must be realized that multidimensional analysis necessitates two additional steps. First, the characteristics formulation must be extended to accommodate strong discontinuities. Second, the characteristics numerical integration must be confined to each region of influence encountered in the solution domain. For nonlinear material motions a third step is essential consisting of an all-events method of integration that mutates along the actual existing waves. Only then is adequate resolution reached for multidimensional wave propagation problems; the numerical results are then intrinsically stable and computer usage is minimal. Since, in the resultant deformation, the various wave fronts are explicitly revealed, physical interpretation is ensured. The characteristics approach, well posed, readily lends itself in a compatible manner to nonlinear transient deformation where strong shock waves occur.

3:36

N9. Experimental study of streaming flows around a levitated sample in an ultrasonic standing wave. E. H. Trinh (Jet Propulsion Laboratory, California Institute of Technology, Pasadena, CA 91109)

Qualitative and quantitative investigations of the streaming flows caused by high intensity ultrasonic fields have been carried out in order to characterize the environment around levitated solid and liquid samples. The interaction of these second-order convective flows with the sample is of primary interest in the case where the properties of the levitated materials are under investigation. The detailed flow characteristics have been found to depend on the boundaries defining the ultrasonic standing wave and the enclosure around the sample region. Results of flow visualization experiments and velocity measurements will be presented for different frequencies and boundary geometries. [Work supported by NASA.]

3:48

N10. Numerical computations of rectified diffusion in nonlinear bubble oscillations. J. B. Fowlkes (National Center for Physical Acoustics, University, MS 38677) and Andrea Prosperetti (Department of Mechanical Engineering, Johns Hopkins University, Baltimore, MD 21218 and National Center for Physical Acoustics, University, MS 38677)

A previous approach to the problem of rectified diffusion in bubble oscillations [J. Acoust. Soc. Am. **37**, 493–503 (1965)] has been applied to

TUESDAY PM

the recent nonlinear bubble dynamics theory of Prosperetti *et al.* [J. Acoust. Soc. Am. **83**, 502–514 (1988)]. A numerical solution is given that calculates growth rates and pressure amplitude thresholds for rectified diffusion and allows modification of the radial oscillations of bubbles due to gas diffusion. The results of the new approach are compared to other rectified diffusion theories, and differences in radial bubble oscillations with and without diffusion are examined. Some experimental measurements of rectified diffusion are also compared to the theoretical results. [Work supported by NIH.]

4:00

N11. Experimental measurements of rectified diffusion in oscillating bubbles. R. G. Holt and L. A. Crum (National Center for Physical Acoustics, University, MS 38677)

An optical scattering technique previously reported on [J. Acoust. Soc. Am. Suppl. 1 **82**, S12 (1987)] has been used to measure growth rates and pressure thresholds for rectified diffusion in oscillating bubbles as a function of the equilibrium radius of the bubbles. Using an Ar–I laser, the scattered light intensity from bubbles with sizes ranging from 20–80 μ and driven by a 25-kHz sound field is measured and used to determine the bubble radii as a function of time. The measurements are compared to previous experimental measurements [J. Acoust. Soc. Am. **68**, 203–211 (1980)], and also to recent numerical results. [Work supported by ONR.]

4:12

N12. A study of the timing of sonoluminescence flashes from stable cavitation. D. F. Gaitan, L. A. Crum, and C. C. Church (National Center for Physical Acoustics, University, MS 38677)

In a previous paper [J. Acoust. Soc. Am. Suppl. 1 **80**, S24 (1986)], experimental measurements of the timing of sonoluminescence (SL) flashes from stable cavitation in gassy water were described. The theory has been extended to examine larger acoustic pressure amplitudes and excellent agreement has been found between the measured and predicted phase angles of the emitted flashes. Since this phase angle is sensitive to the explicit formulation of the equations for bubble dynamics, these data can be used to examine the applicability of various theoretical models. The results of this comparison will be presented and reasons for the inapplicability of certain formulations will be given. [Work supported in part by the NIH and the ONR.]

4:24

N13. A comparison of the thresholds for free-radical generation and transient cavitation activity induced by short pulses of ultrasound. Ronald A. Roy and J. B. Fowlkes (National Center for Physical Acoustics, University, MS 38677)

Insonification of relatively dirty water with short pulses (order 10 μs) of high-frequency (1-MHz) focused ultrasound can lead to acoustic cavitation. If the bubble activity is sufficiently energetic, free radicals are produced that can be detected as sono-or chemiluminescence in the fluid. The question remains as to how the thresholds for light production compare to measured thresholds for transient bubble activity. The threshold for chemiluminescence in gassy water doped with luminol using photon counting techniques was measured. In addition, thresholds were *simultaneously* monitored for bubble activity using a high-frequency (30-MHz) acoustic backscattering technique. All measurements were carried out for various combinations of acoustic pulse length and repetition frequency, and were compared with previous investigations as well as theoretical predictions. [Work supported in part by ONR and NIH.]

4:36

N14. Thermoacoustic generation of narrow-band signals with high repetition rate pulsed lasers. Yves H. Berthelot (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

The generation of sound by absorption of laser light in water is analyzed for situations where the laser is pulsating at a high repetition rate. It is shown that pulsating the laser at any arbitrary repetition rate is, in general, not a very effective way to produce a narrow-band signal. For narrow-band sound generation, one should instead pulsate the laser at an optimum repetition rate that is determined by the optical frequency of the laser and, to a lesser extent, by the laser beam diameter. Expressions for the optimum repetition rate are derived from both a frequency domain analysis and a time domain analysis. It is found that, with the present laser technology, it is now possible to generate continuous thermoacoustic directional sound waves with high repetition pulsed lasers in such a way that these signals are detectable several kilometers away from the source. Spatially periodic laser deposition configurations on the water surface are also discussed, and further improvement in signal-to-noise ratio can be achieved, in principle, for a spatial periodicity tuned to the optimum temporal periodicity of the repetition rate of the pulsed laser.

4:48

N15. Acoustically generated temperature gradients in short plates. Anthony A. Atchley, T. J. Hofler, and M. D. Kite (Physics Department, Naval Postgraduate School, Monterey, CA 93943)

The results of measurements of acoustically generated temperature gradients in short, thin plates located in a resonant tube were reported previously [M. Muzzerall *et al.*, J. Acoust. Soc. Am. Suppl. 1 **82**, S21 (1987)]. The previous measurements agree well with theory [J. Wheatley *et al.*, J. Acoust. Soc. Am. **74**, 153–170 (1983)] as long as the acoustic pressure amplitude is less than or equal to 150 dB *re:* 20 μPa. Measurements made at a pressure amplitude of 162 dB show serious disagreement with theory in regions of high acoustic velocity. In order to investigate this disagreement further, a more extensive set of measurements was performed, employing various mean pressures, acoustic pressure amplitudes, plate lengths, and plate configurations. The results of these measurements are reported. The data are compared to theory with particular emphasis on velocity-dependent effects. [Work conducted for the Office of Naval Research and funded by the Naval Postgraduate School.]

5:00

N16. Static pressure differential measurements in a thermoacoustic engine. T. Hofler (Department of Physics, Naval Postgraduate School, Monterey, CA 93943)

High-amplitude standing waves interact with the solid boundaries of a resonator in the region of the acoustic boundary layer. This interaction causes thermoacoustic heat transport as well as a partial rectification of the oscillatory velocity into a steady flow, known as acoustic streaming. The traditional analysis of streaming assumes that the bore of the resonator duct is much larger than the boundary layer and that there is no net flow through the closed vessel, which requires that a counter flow at the center opposes the streaming flow at the walls. However, the "stack" of a thermoacoustic engine has channels that are only two to four boundary layers wide. Any streaming counter flow in these tiny channels will generate substantial viscous shear that must be sustained by a static pressure gradient. The results of measurements of static pressure differences across the "stack" will be presented.

5:12

N17. Seawater as a working fluid for thermoacoustic engines. G. W. Swift and A. M. Fusco (Condensed Matter and Thermal Physics Group, Los Alamos National Laboratory, Los Alamos, NM 87545)

In thermoacoustic prime movers (the modern embodiment of the Sondhauss tube), a temperature gradient imposed on the fluid in a resonator produces acoustic power at the resonance frequency. Such sound sources have been built using air, helium gas, and liquid sodium as working fluids. A complete lack of moving parts makes thermoacoustic sound sources simple and reliable. The thermodynamic and transport properties of water are presented with emphasis on water's virtues as a thermoacoustic working fluid, and the possibility of using a water thermoacoustic engine as a high power SONAR projector is discussed. [Work supported by DOE/BES.]

5:24

N18. Performance of a liquid-sodium thermoacoustic engine. A. Migliori and G. W. Swift (Condensed Matter and Thermal Physics Group, Los Alamos National Laboratory, Los Alamos, NM 87545)

A thermoacoustic engine that uses liquid sodium as its working substance has been constructed. The engine generates acoustic power in liquid sodium, using heat flowing from a high-temperature source to a low-temperature sink. The measured performance of this engine disagrees significantly with numerical calculations based on a linear theory of thermoacoustic engines. The efficiency of the engine is a substantial fraction of Carnot's efficiency, and its power density is comparable to that of the conventional heat engines in widespread use. Thus this type of engine may ultimately be of practical, economic importance. [Work supported by DOE/BES.]

5:36

N19. Acoustically driven instability of a liquid drop under microgravity. Martin Manley and Vineet Mehta (Department of Electrical Engineering, Laboratory for Advanced Computation, University of Lowell, 1 University Avenue, Lowell, MA 01854)

When the media surrounding a heated liquid drop is excited by an acoustic wave, acoustic streaming can modify the transfer of heat from the drop's surface. Spatial variations in the heat transfer can result in an adverse surface tension gradient and as well as internal flows in the drop. These flows can also result in unstable modes of vibration. The stability of surface waves on a liquid drop suspended by a plane acoustic wave in a microgravity environment will be discussed. In particular, how acoustic streaming affects the stability and heat transfer at the liquid–gas interface will be addressed.

5:48

N20. The effects of vibration on the flow of a non-Newtonian fluid. W. Jack Hughes and Andrew N. Vavreck (Applied Research Laboratory, Pennsylvania State University, P. O. Box 30, State College, PA 16804)

The flow characteristics of non-Newtonian fluids are nonlinear and depend upon the generation of a minimum or threshold shear stress before flow is initiated. These fluids are composed of aggregate solids suspended in a fluid medium; examples are concrete and printer's ink. A non-Newtonian fluid was vibrated with ultrasonic and low audio frequencies while its flow characteristics relative to the intensity of vibration were noted. The frequency–amplitude product for threshold shear stress was investigated and has been replaced by a function that is more highly frequency dependent.

TUESDAY AFTERNOON, 15 NOVEMBER 1988 — MOLOKAI ROOM, 2:00 TO 3:00 P.M.

Session O. Physiological Acoustics II: Ototoxicity of Environmental Chemicals

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Invited Papers

2:00

O1. Chemical asphyxiants as sources of ototoxicity. Laurence D. Fechter (Department of Environmental Health Sciences, Johns Hopkins University, Baltimore, MD 21205)

Total asphyxiation has repeatedly been shown to disrupt cochlear function, but moderate reduction in oxygen delivery has also been suggested as a general mechanism responsible for auditory dysfunction. Several recent experiments in this laboratory using environmental contaminants to produce moderate hypoxia have been used to study the parameters under which reductions in oxygen delivery can disrupt cochlear function. Both transient and permanent impairments have been studied. Acute carbon monoxide exposure increases cochlear blood flow and at higher doses produces transient disruption of the compound action potential. This loss occurs specifically at high frequencies. When carbon monoxide and noise are presented simultaneously, permanent auditory loss results, which is substantially greater than the effect of noise alone. The impairment is greatest at high frequencies and injury to hair cells occurs in the basal turn when white noise is employed. Evidence of ototoxicity by such chemical asphyxiants as butyl nitrite and trimethyltin will also be presented. Taken as a whole, the data do support the hypothesis that environmental agents which impair oxygen delivery and utilization are potent ototoxicants and suggest that hypoxia may be an important mechanism for disrupted auditory function.

2:16

O2. Otoneurologic manifestations following carbon monoxide poisoning. Kazumi Makishima (Department of Otorhinolaryngology, Kyushu University, School of Medicine, Fukuoka, 812 Japan)

Otoneurologic examinations were conducted in the 15 patients who suffered from acute carbon monoxide (CO) poisoning in a coal mine accident. The audiologic and vestibular derangement found in the patients were not significant, but were suggestive of central origin. One victim in the same accident, who had severe psychopathological symptoms and had no otoneurologic complaint, showed moderate histopathological changes in the cerebrum in the presence of no specific change of the inner ears. The effect of CO poisoning on the threshold sensitivity of the auditory responses was studied in a CO poisoning animal model of guinea pigs. Loss of auditory threshold sensitivity was most prominent at the auditory cortex, and was next most severe at the inferior colliculus. There was no loss of sensitivity at the cochlea. Some of the patients mentioned above have been studied again about 25 years after the accident. It is felt that there must be increased and more considerable widespread pathology in the brain following CO poisoning to explain the present audiometric and vestibural pictures.

2:32

O3. Effects of lead acetate on guinea pig—Electrophysiological study of the cochlear function. K. Yamamura, H. Ohno, K. Terayama, N. Yamamoto, and R. Kishi (Department of Hygiene, Asahikawa Medical College, Asahikawa, 078 Japan)

Segmental demyelination and axonal degeneration of motor nerves induced by lead exposure are well known in man and animals. The effect of lead acetate exposure to man may involve the cranial nerves, since vertigo and sensory neuronal deafness have been reported among lead workers. However, there are only a few reports concerning the dose effects of lead acetate both on the cochlear and the cranial VIII nerve. The effects of lead acetate on the cochlea and the VIII nerve using CM (cochlear microphonics), AP (action potential), and EP (endocochlear potential) of the guinea pigs are investigated. The results of this experiment were as follows: No changes of CM and EP were found by lead acetate exposure. On the other hand, there were an elevation of threshold but a decrease of maximum output voltage on AP.

2:48–3:00
Discussion

Session P. Speech Communication III: Special Focus Session I: Speech Processing Aids for the Handicapped (Lecture and Poster Session)

A. Maynard Engebretson, Cochairman
Central Institute for the Deaf
818 S. Euclid
St. Louis, Missouri 63110

Shizuo Hiki, Cochairman
School of Human Sciences
Waseda University
Mikajima, Tokorozawa, 359 Japan

Chairman's Introduction—2:00

Invited Papers

Please note: The first three papers in this session will be presented in lecture format. Papers PP4-PP34 will be presented as posters. All posters should be set up before 2:00 p.m. All posters will be displayed from 2:00 to 6:00 p.m. To allow contributors the opportunity to see other posters, contributors of even-numbered papers will be at their posters from 3:30 to 4:30 p.m. and contributors of odd-numbered papers will be at their posters from 4:30 to 5:30 p.m.

2:05

P1. Signal processing for sensory aids. Harry Levitt (Center for Research in Speech and Hearing Sciences, City University of New York, 33 West 42nd Street, New York, NY 10036)

Sensory aids can be subdivided in two important ways: by modality (auditory, visual, tactile, or direct electrical stimulation) and by degree of signal processing (nonspeech, speech-specific, feature-extraction, and speech-recognition). Nonspeech processing aids are designed to make maximum use of the impaired sensory system regardless of whether communication is by speech or other means. Speech-specific processing is designed to match the average spectral and temporal characteristics of the speech signal to the characteristics of the impaired auditory system. Feature-extraction systems involve the automatic extraction of phonetic or articulatory features from the speech signal. Speech-recognition processing makes use of automatic speech recognition techniques to facilitate the communication process. The application of each of these different forms of signal processing to sensory aids of various kinds will be described. [Work supported by NINCDS.]

2:25

P2. Enhancing the clarity of speech for listeners with hearing impairments. Louis D. Braida (Research Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge, MA 02139)

Many listeners with hearing impairments exhibit reduced auditory reception of speech both in quiet and in the presence of interfering sounds. A variety of efforts intended to enhance the clarity of speech for such listeners are reviewed. The results obtained will be interpreted in terms of a model [P. M. Zurek and L. A. Delhorne, J. Acoust. Soc. Am. **82**, 1548–1559 (1987)] that treats the hearing loss as equivalent to an additive noise with a power spectrum determined by the audiogram. [Work supported by NIH.]

2:45

P3. Computer-based speech training aids. Charles S. Watson and Diane Kewley-Port (Department of Speech and Hearing Sciences, Indiana University, Bloomington, IN 47405)

During the past 20–25 years, a number of research groups, and also individual investigators and clinicians, have attempted to apply computer technology to the problem of teaching speech to deaf or misarticulating children. Most early computer-based training aids used single- or multidimensional visual displays of speech as alternatives to auditory feedback. More recently, computer-based speech recognizers have provided evaluations of the "quality" of utterances, eliminating the requirement that the child interpret various complex transforms of the acoustic waveform. In other systems, feedback is based on the output of sensors that measure the position or motion of the articulators. Positive results obtained with several of these approaches suggest that this line of research will lead to common use of computer-based speech training within 5–15 years. [Research supported by NSF.]

3:05–3:15

Break

3:15–3:30

Chairman's Comments

Invited Poster Papers

P4. Noise reduction strategies for the hearing aid user—A critical review. Ora Buerkli-Halevy (PHONAK AG, Staefa, Switzerland)

As recently confirmed by surveys conducted by the Federal Trade Commission and the Hearing Industry Association, communication in noise is still the number one problem of hearing aid users. This is confirmed by many studies which concluded that SNR enhancement is the single most important determinant of perceived benefit from amplification. In the last few years, development of special noise reduction circuitry has been a priority for many hearing instrument manufacturers. The resulting design can be divided into three major categories: automatic low-frequency suppression, user operated noise reduction systems, and adaptive compression systems. In order to make appropriate fitting choices, it is important to understand how these different systems function and in which environment each is most effective. A critical analysis of each signal processing scheme and its effect on speech discrimination in noise of each system is therefore essential.

P5. Evaluation of a digital hearing aid and fitting system. Michael P. O'Connell, Margaret W. Skinner, A. Maynard Engebretson, James D. Miller, and David P. Pascoe (Central Institute for the Deaf, 818 S. Euclid, St. Louis, MO 63110)

The results of the evaluation of a body-worn, programmable digital hearing aid in short-term field studies of 3 days to 3 weeks on six hearing-impaired subjects will be presented. The hearing aid was programmed to implement a four-channel instantaneous compression hearing aid model. Frequency selective channel gains and maximum power output settings of the hearing aid were controlled by a computer-based, hearing-aid evaluation system. This system integrated real-ear measurements and hearing aid control capabilities to achieve several different real-ear frequency/gain characteristics for each subject. The desired real-ear gains were calculated based on a prescriptive fitting method applied to hearing threshold and most comfortable listening levels. The performance of the device with the user-preferred real-ear gain curve was evaluated using speech intelligibility testing and HAPI questionnaires completed after using the digital hearing aid. The speech intelligibility testing consisted of a computer automated test using the CUNY nonsense syllables for seven different speech levels under three noise conditions (21 total conditions). All subjects were current hearing aid users and completed identical intelligibility tests and HAPI questionnaires with their own hearing aids. [Work supported by the VA, the 3M Corporation, and NASA.]

P6. Comparison of speech recognition results for a one- and two-formant speech coding strategy for a multichannel cochlear implant. Judith A. Brimacombe, Anne L. Beiter, Mary J. Barker, Karen A. Mikami, and Steven J. Staller (Cochlear Corporation, 61 Inverness Drive East, Englewood, CO 80112)

This study compared clinical results obtained from 96 postlinguistically deafened adults implanted with the Nucleus 22 channel cochlear implant system. Seventy-one patients were fitted with a device that tracked the second formant of the voice, while 25 used one that tracked the first and second formant frequencies. Experience with the device averaged 3.7 months. Four subtests (four-choice spondee, vowel, NU#6, and CID sentences) of the Minimal Auditory Capabilities Battery (Auditec of St. Louis recording) were used. In addition, 209 patients were administered live-voice measures (continuous discourse tracking and vowel/consonant identification) in three conditions: lipreading, lipreading plus device, and device only. Results revealed that the group using the two-formant tracker performed significantly better than those using the one-formant tracker on the following measures: (1) NU#6 monosyllabic word test; (2) CID sentences; (3) continuous discourse tracking difference score (lipreading plus device minus lipreading); and (4) vowel and consonant identification in the device only condition. These findings suggest greater efficacy with the two-formant tracker; this is the strategy commonly in clinical use at the present time.

P7. Speech perception with cochlear implants and tactile aids. P. J. Blamey, G. M. Clark, and R. C. Dowell (Department of Otolaryngology, University of Melbourne, Royal Victorian Eye and Ear Hospital, 32 Gisborne Street, East Melbourne 3002, Australia)

Three related areas in our research are speech perception in noise, consonant recognition, and multimodal speech perception. Perception of speech in four-speaker babble was evaluated with a multiple channel cochlear implant presenting first- and second-formant information. In quiet, scores were 80% for vowels, 48% for consonants, and 51% for sentences. The signal-to-noise ratios required to reduce scores to 75% of the value in quiet were 8 dB for vowels, 12 dB for consonants, and 14 dB for sentences. Detailed consonant studies showed that amplitude information contributed directly to the recognition of voicing and manner of articulation and influenced the perception of other cues. In addition to auditory cues, lipreading is commonly used with implants and other aids. A probabilistic model of feature recognition describes the combination of auditory plus visual information well. The combined effect of tactile information is overestimated by the model for minimally trained subjects. The model may clarify issues in the training of hearing-impaired children, as well as the design of speech processing aids.

TUESDAY PM

P8. Within-subject comparisons of analog and pulsatile speech processors for cochlear implants. Sigfrid D. Soli (Hearing Research Laboratory, Building 270-4S-11, 3M Center, St. Paul, MN 55144) and Blake S. Wilson (Neuroscience Program, Research Triangle Institute, Research Triangle Park, NC 27709)

Analog and pulsatile speech processors for cochlear implants represent widely different strategies for coding perceptual information in speech. Between-subject comparisons of their efficacy, however, have been confounded by large and unquantified subject differences. Two patients implanted with multiple electrode arrays have been tested both with four-channel analog and pulsatile coding strategies. The analog processor performed bandpass filtering with syllabic compression/limiting and stimulated four bipolar electrode pairs. The pulsatile processor delivered volleys of biphasic pulses at the rate of $F0$ during voiced intervals of speech to four of six possible bipolar electrode pairs selected on the basis of spectral peak frequencies. Tests of auditory capabilities showed that the two coding strategies provided similar overall benefit. Analyses of consonant and vowel confusions revealed that the two strategies were not equally effective for some types of speech information. The results of these analyses can be used to identify the strengths and weaknesses of analog and pulsatile speech coding strategies for cochlear implants.

P9. A speech coding method for an auditory prosthesis by extracochlear stimulation. Tohru Ifukube, Yoshihiro Hirata (Research Institute of Applied Electricity, Hokkaido University, N12-W6, Kita-ku, Sapporo, 060 Japan), and Jun'ichi Matsushima (Medical School, Hokkaido University, N15-W7, Kita-ku, Sapporo, 060 Japan)

A new model of auditory prosthesis using a newly developed electrode is proposed. This prosthesis is less invasive than conventional ones because it can be placed superficially over the round window membrane of the cochlea rather than being inserted into the scala tympani. The electrode is coated with polyvinyl alcohol gel and is suitable as an extracochlear prosthesis because the electrode can firmly adhere to the round window membrane without damaging it. As a speech coding method, a new idea has been proposed to simultaneously transmit the pitch signal and the second formant frequency through the electrode. In this method, an additional stimulation pulse is inserted between successive pitch pulses, and the time delay between the additional pulse and the preceding pitch pulse is set in proportion to the second formant frequency. Two digital signal processors are used to extract the pitch and the second formant frequencies in real time. These time sequential stimuli are used for deaf patients. This new method of speech coding has proven effective for discrimination of the five Japanese vowels, and also for discrimination among pitch patterns.

P10. Speech processing to aid the profoundly deaf. Georges Vilaclara[a] (EPFL, DE-LEMA, CH-1015 Lausanne, Switzerland)

This work is part of an extended project to transmit speech information to the profoundly deaf through an alternate sensory modality. A tool was developed to select and define codes to complement speechreading. The learning procedure operates on a set of speech waveforms (known database), preprocessed to produce a large collection of parameters at regular time intervals, which is phonetically segmented according to a stability function. The experimentalist identifies segments in terms of units (e.g., linguistic, acoustic, or articulatory units). For each unit, the variability of its parameters is automatically represented by statistical patterns. In addition, probabilistic rules describing the unit's characteristics and properties can then be defined. All these specifications are stored in a knowledge base. Two techniques are used to recognize the units in unknown speech. A pattern matcher determines the most suitable statistical pattern to represent the current frame. An expert system supervises the pattern matcher, analyses results, and/or infers decisions. Presented in a suitable way (user defined) via a prosthesis, the code can be tested to determine its utility to cross-modal speech perception. [Work supported by the Swiss NSF.] [a] Current address: Center for Auditory and Speech Sciences, Gallaudet University, Washington, DC 20002.

P11. Visual recognition of words from palatometric displays. Samuel G. Fletcher (Department of Biocommunication, University of Alabama at Birmingham, Birmingham, AL 35294)

The relation between visually perceived palatometric patterns and word recognition was measured using palatometric representations from 16 words shown on a video monitor. Five subjects examined the patterns and identified the words spoken. Words identified at 90% or higher accuracy rate were classified as "recognized." Virtually immediate recognition of some words and rapid development of recognition for all of the words were found. Few errors were observed in the responses of any subject after the 6th of 10 experimental days. Analysis of the data indicated that word length, consonant manner of articulation, consonant and vowel place of articulation, and movement timing were the visual attributes used to differentiate the palatometric displays. An articulatory feature analysis indicated that the visually perceived attributes of the displays parallel those found in acoustic studies. The findings suggest that phonetic features may not be sensory avenue dependent. The findings will be interpreted with respect to modeling and shaping speech of the deaf. [Work supported by the NINCDS.]

P12. The Indiana speech training aid (ISTRA). Diane Kewley-Port, Charles S. Watson, and Mary Elbert (Department of Speech and Hearing Sciences, Indiana University, Bloomington, IN 47405)

The ISTRA system is being developed to improve the speech of deaf or misarticulating children. The system consists of a microcomputer equipped with a low-cost, speaker-dependent speech recognizer. One-line feedback concerning the *quality* of entire utterance is derived from the degree of match between a stored template and the new utterance. Software for conducting speech training (under the guidance of a speech pathologist) has been developed over 3 years based on the collaboration of research scientists, clinicians, and programmers. Clients are able to run their speech drills independently using a variety of drills in game formats. Ongoing clinical evaluation of ISTRA training has employed single-subject design experiments. In these studies, "blind" listener juries rated the speech of words collected before, during, and after training. Results show significant improvement of the quality of words trained in the ISTRA drill, as well as generalization to nontrained words. One ISTRA system has been used for regular speech training in an elementary school with positive results, both for improved speech as well as enthusiastic acceptance of the device. A description of the results of 3 years of research with ISTRA will be presented. [Research supported by NSF.]

P13. Speech visualization and its application for the hearing impaired. Akira Watanabe and Yuichi Ueda (Department of Electrical Engineering and Computer Science, Kumamoto University, Kumamoto, 860 Japan)

A speech-visualization system has been developed as an extension of research on a color display system for connected speech [Watanabe *et al.*, IEEE Trans. Acoust. Speech Signal Process. **ASSP-33**, 164–173 (1985)]. In this newly developed system, normalized spectrographic patterns are overlapped as a change of luminance on a whole pattern that represents voiced segments by colors. The spectrographic patterns are represented with high luminance in unvoiced portions, and with low luminance in voiced ones so as not to disturb the color image. This extension allows unvoiced plosives as well as fricatives and affricates to be clearly visualized by the sharp contrast between their black-and-white patterns and the vivid colors of the voiced components. Moreover, in order to generate a visually clear image, a cleaner, which eliminates unnecessary patterns such as aspirates that are often observed at the end of utterances, is introduced. Finally, two applications of this system are described. One is a speech training system in which a personal computer, with a database, controls the speech-visualization system and provides a hearing aid. The other is a telecommunication system that utilizes a public telephone channel. [This work was supported by TAF.]

P14. A new speech training system for profoundly deaf children. Yoshinori Yamada, Norio Murata (Central Research Laboratories, Matsushita Electric Industrial Co., Ltd., Moriguchi, 570 Japan), and Tatsuo Oka (School for the Deaf, University of Tsukuba, Konodai, Ichikawa, 272 Japan)

A new speech training system based on a personal computer has been developed. Monitoring of the articulatory movements and speech sounds of individuals was accomplished using five sensors: a microphone, a neck sensor for vocal cord vibration, a nose sensor for nasal sound, an expiratory airflow sensor, and a tongue position sensor. Signals from these sensors were processed in real time by a DSP to extract ten training parameters that are useful for prosodic or phonemic training. These parameters were transferred to an MSX personal computer every 10 ms and were displayed on a color CRT screen together with a training model in front of which the trainee repeatedly practices his speech. Such training was conducted for profoundly deaf children (aged 4–6) for 2 years at the School for the Deaf, University of Tsukuba, and has been enthusiastically accepted and used. After 2 years of this training, remarkable improvements in the intelligibility of monosyllables were attained that had not been attained by conventional methods.

TUESDAY PM

P15. A sentence-level speech training system for hearing-impaired children. Minoru Shigenaga and Hirooka Muramatsu (Faculty of Engineering, Yamanashi University, 4 Takeda, Kofu, 400 Japan)

Hearing-impaired children, or even adults, cannot necessarily pronounce words or sentences fluently, even though they can clearly articulate individual vowels. The system described here aims at helping teachers train hearing-impaired children by showing the differences between the children's pronunciations and the correct ones given by teachers. The system displays the speech waveform uttered by a trainee, together with several useful correlates including frequency spectra, vocal tract area functions, and the phonetic symbols for estimated vowels. The lateral shapes of the vocal tract are also estimated for vowel segments and displayed together with the correct shapes so that the trainee can practice the correct articulation of vowels referring to the place of articulation and the lip opening. The fundamental frequency contour and the waveform envelope of the trainee's speech are also displayed together with the teacher's, so that the trainee can practice pronunciations with correct prosody. This system is designed so as to be used also for self-training by older children with hearing impairment.

P16. Linguistic processing in aids for the handicapped. Sheri Hunnicutt (Department of Speech Communication and Music Acoustics, Royal Institute of Technology, S-100 44 Stockholm, Sweden)

Since 1971, various linguistic components have been developed for text-to-speech systems that are used as aids for disabled persons, for symbol-to-speech systems for nonspeaking individuals, and for word prediction for motorically disabled persons. The text-to-speech systems, or later versions of them, are used as speech prostheses for nonvocal persons and as computer display readers for blind computer users. The symbol-to-speech system is an expanded version of one of the text-to-speech systems [Carlson *et al.*, Proc. ICASSP (1982)]. To convert symbols to phoneme strings, both lexicons and parsers were developed for four languages. In two languages, symbol-to-text systems were also developed. These systems are used as communication aids by persons experiencing severe speech and motor disabilities. Several algorithms have been designed to predict words from partial information. One program is being used as a speaking or writing aid. A second program is being tested as an aid in aphasia rehabilitation. It is planned that the third algorithm will be used by speaking persons in need of written output capability in conjunction with a speech recognition system.

P17. An integrated voice analysis system for clinical use. Hideki Kasuya and Yoshinobu Kikuchi (Faculty of Engineering, Utsunomiya University, 2753 Ishii-machi, Utsunomiya, 321 Japan)

The need for acoustic methods for the quantitative evaluation of voice quality arises in many voice research and clinical areas. Since the voice quality is associated with multidimensional physical characteristics of a voice signal, the acoustic evaluation must be made on the basis of various acoustic measurements from the signal. An integrated voice analysis system has been developed, primarily aiming at applications for the quantitative evaluation in voice clinics. The system uses the ARIEL DSP-16 and an IBM-PC/AT computer. The acoustic measurements made by the system include: the average pitch frequency, jitter, shimmer, vocal noise, gross spectral shape, formant frequencies and levels, temporal variations in the frequency spectrum, and so on. The key features of the system and several examples of its clinical application will be presented.

P18. An analysis of the characteristics of speech perception ability by the combined use of cochlear implant and lipreading. Yumiko Fukuda (Research Institute, National Rehabilitation Center for the Disabled, Tokorozawa, 359 Japan)

In this study, test materials for evaluating speech perception ability by a combined use of cochlear implants and lipreading were first selected, taking into account various acoustical parameters such as fundamental frequency, intensity, and formant frequencies that can be transmitted through a speech signal processor in a cochlear implant, as well as supplemental speech information that can be obtained through mouth shape in lipreading. The test materials included the short versus long vowels, unvoiced versus voiced consonants. Japanese word accent and sentence intonation, and the 5 Japanese vowels, 50 monosyllables, 50 words, and 29 sentences that consisted of 100 words. Then, both the speech signal and the mouth shape for utterances of the test materials by a female speaker were recorded on videotapes and were presented to the subjects who had cochlear implants in Tokyo. The results showed that the test materials were useful for a detailed analysis of the characteristics of speech perception ability through the cochlear implants and lipreading.

P19. Artificial speech production device for laryngectomees controlled by jaw and tongue movements. Masafumi Matsumura (Department of Computer Science and Systems Engineering, Faculty of Science and Engineering, Ritsumeikan University, Tojiin, Kita-ku, Kyoto, 603 Japan), Yoshiyuki Goi, and Katsuhiko Fujii (Department of Electrical Engineering, Faculty of Engineering, Osaka University, Yamada-oka, Suita, 565 Japan)

The purpose of this study is to develop an artificial speech production device for laryngectomees. The device is composed of three parts, a measurement part for jaw and tongue movements, an estimation part for the formant frequencies, and a voice synthesizer. This paper argues that the formant frequencies of vowels can be estimated from tongue and jaw movements of the subject, and that based on this information, continuous vowels can be produced using a voice synthesizer. Curvature of midsagittal tongue surface is measured with piezoelectric film. The degree of the jaw opening is estimated by the electrical impedence between right and left buccalis. Experimental results suggest that the formant frequencies can be determined from the curvature of the tongue surface and the jaw opening. Therefore, a mapping function is introduced to obtain the formant frequencies for the synthetic speech. The amplitude and pitch of the voice are controlled manually by the person who wears the device. The articulation score of the voice produced by the device was 95% for isolated vowels and 90% for connected vowels. Results showed the usefulness of the proposed device.

P20. Design of an intelligent artificial larynx. Shizuo Hiki (School of Human Sciences, Waseda University, Tokorozawa, 359 Japan) and Yuki Kakita (Department of Electronics, Kanazawa Institute of Technology, Kanazawa-Minami, 921 Japan)

The artificial larynx proposed in this paper can control the following features automatically: (a) voicing onset/cessation, (b) a brief stop in vibration for voiceless consonants, (c) variation in fundamental frequency, and (d) intensity for prosodic information. This device consists of three parts: (1) a detector of biological signals, i.e., sound, electromyograms (surface), air pressure and airflow, tactile information on the skin surface surrounding the larynx, etc.; (2) the controller for sound source, in which control parameters are derived from the biological signals; and (3) a sound source, which is to be implanted in the neck. To investigate the basic characteristics of the timing factors, (a) and (b) mentioned above, a manual operation scheme was examined by employing specially designed switches. The timing control is to be replaced by the remaining functions of the speech muscles. Also, a result of the experiment on the control of prosodic information, (c) and (d) above, using airflow, is presented. [Work supported by Grant-in-Aid for Scientific Research, The Ministry of Education, Science and Culture, Japan.]

Contributed Poster Papers

P21. Predicting aided speechreading performance in segment identification tasks. Louis D. Braida (Research Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge, MA 02139)

Auditory presentation of low-frequency speech components, and tactile or electrocochlear stimulation derived from speech, can improve the speechreading of speech segments. A method is described for predicting both the level of aided performance and the resulting error pattern from the confusion matrix for each modality separately. The method is based on a characterization of confusion matrices in terms of a multidimensional Thurstonian decision model that allows performance to be described in terms of sensitivity and bias. In the multimodal case, the deci-

sion space is assumed to be the product space of the decision spaces corresponding to the stimulation modes. In certain cases, multimodal sensitivity is roughly equal to the vector sum of the sensitivities for each stimulation mode, indicating that cues are integrated with little interference. [Work supported by NIH.]

P22. Comparison of three sentence-level tests for evaluating audiovisual performance of subjects using a cochlear implant. W. M. Rabinowitz, Ken W. Grant, and D. K. Eddington (Research Laboratory of Electronics, Massachusetts Institute of Technology, Room 36-757, Cambridge, MA 02139)

Postlingually deafened adults implanted with the Symbion cochlear prosthesis have been tested for audiovisual reception of isolated sentences. Three different sets of sentence materials have been used: (a) CID everyday sentences, including live and recorded versions; (b) CUNY sentences developed by Boothroyd *et al.* (CUNY Rep. RCI10), presented from videodisc; and (c) IEEE/Harvard sentences, from in-house recordings. The tests were conducted in three modes: vision (speechreading) alone, sound alone, and sound plus vision. No feedback was given and, except for the live-voice CID tests, sentences were never repeated for a given subject. Results indicate that all three materials are sensitive to detecting whether the cochlear implant provides a benefit to speechreading, but the CID and CUNY sentences frequently exhibit ceiling effects that limit their ability to differentiate the degree of benefit. The more challenging IEEE/Harvard sentences have reduced ceiling effects and provide other useful performance measures. Specifically, the benefit to speechreading provided by the prosthesis at speech-to-noise ratios of 0–3 dB is roughly one-half that provided in quiet. [Work supported by NIH.]

P23. Amplitude envelopes and speechreading. Ken W. Grant, Rebecca J. Renn, and Jenny S. Yu (Research Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge, MA 02139)

Amplitude envelopes derived from filtered bands of speech provide varying degrees of benefit to speechreading. Three parameters related to envelope extraction are examined using both easy and difficult sentence materials: (1) the bandwidth and center frequency of the filtered speech signal used to obtain the envelope, (2) the bandwidth of the envelope signal, and (3) the carrier signal used to convey the envelope. Thus far, results for normal hearing subjects presented with difficult speech materials suggest that: (1) the envelope of wideband speech did not always provide the greatest benefit to speechreading when compared to envelopes derived from octave bands of speech; (2) as the bandwidth of the envelope signal increased from 100–4000 Hz, auditory–visual (AV) performance improved slightly, whereas bandwidths below 100 Hz resulted in AV scores that were equal to or worse than speechreading alone; and (3) low-frequency carrier signals were better than high-frequency or wideband carrier signals, at least for envelopes derived from an octave band of speech centered at 500 Hz. Similar tests with easy speech materials are in progress. [Work supported by NIH.]

P24. An application of the articulation index to hearing aid fitting. Christine M. Rankovic and Dianne J. Van Tasell (Department of Communication Disorders, University of Minnesota, Minneapolis, MN 55455)

It is often assumed that the articulation index (AI) is a sensitive and appropriate metric for making comparisons among frequency-gain characteristics of hearing aids, and that the characteristic yielding the highest AI will result in the best speech intelligibility score. In evaluating these assumptions, 12 hearing-impaired subjects listened to nonsense syllables under conditions of amplification prescribed by the NAL [D. Byrne and H. Dillon, Ear Hear. **7**, 257–265 (1986)] and POGO [G. A. McCandless and P. E. Lyregaard, Hear. Instrum. **34**, 16–21 (1983)] hearing aid fitting formulas. These two prescriptive methods recommend similar frequency responses, although the POGO consistently provides more overall gain. In a third condition, the goal of amplification was to insure that the speech was fully audible (AI = 1.0). Results suggest that the articulation index is useful for predicting percent-correct intelligibility scores for some, but not all, of the hearing-impaired subjects. Interestingly, the AI = 1.0 condition did not necessarily yield the best percent-correct intelligibility score. [Work supported by NICHHD T32 HD-07151, NINCDS NS 12125, and the Bryng Bryngelson Communication Disorders Research Fund at the University of Minnesota.]

P25. Quasiarticulatory tactile syllables in a long-term training experiment. H. G. Tillmann and H. G. Piroth (Institut für Phonetik und Sprachliche Kommunikation, Schellingstrasse 3, 8000 Munich 40, Federal Republic of Germany)

Tactile syllable equivalents were constructed that map relevant articulatory gestures onto the left forearm in a quasi-isomorphic way [cf. H. G. Piroth, J. Acoust. Soc. Am. Suppl. 1 **79**, S73 (1986)]. The patterns were delivered by the 16-channel "system for electrocutaneous stimulation SEHR-2" [H. G. Tillmann and H. G. Piroth, J. Acoust. Soc. Am. Suppl. 1 **79**, S73–S74 (1986)]. The present investigation was designed to determine whether a subject is able to acquire extensive inventories of tactile patterns. One subject had to train tactile obstruent-vowel syllables with a fixed vowel. During each test run, a subset of the set of tactile obstruents followed by /ɑ:/ was presented repeatedly in a systematic order. Then, the same subset was trained in an identification test with a feedback after each stimulus. When the average recognition rate yielded 80%, the subset was enlarged. After 35 test runs, the subject could identify 12 obstruent patterns (/p,t,k,b,d,g,f,ʃ,x,v,ʒ,v/) at criterion levels. Training the complete set of 16 patterns (/s,z,ç,j/ added), he just failed to reach the criterion after 39 additional test runs (75.8%).

P26. Speech enhancement for the hearing impaired by envelope filtering on a loudness scale. Birger Kollmeier and Volker Hohmann (Drittes Physikalisches Institut, Universität Göttingen, Bürgerstrasse 42-44, D-3400 Göttingen, Federal Republic of Germany)

Dynamic compression systems for sensorineural hearing-impaired subjects reduce intensity fluctuations, although they are an important cue for understanding speech. To overcome this effect, an envelope high-pass filter operating in the loudness domain is evaluated, which even enhances these intensity modulations: The output of a filterbank is low-pass filtered to extract the envelope of incoming speech. Loudness scaling data from normal listeners and from each individual impaired listener is used to convert the envelope to a loudness scale value and to convert the high-pass-filtered loudness back to the "desired" envelope value, respectively. The output signal is finally obtained by summing up the filterbank channels after correcting their amplitude. Loudness scaling and intensity modulation discrimination data of several sensorineural hearing-impaired subjects will be presented, as well as speech intelligibility scores for a German CVC rhyme test with and without the algorithm. First results indicate that even no loudness filtering improves speech intelligibility in a wide range of input levels. However, the effect of high-pass filtering in the loudness domain is only moderate. [Work supported by Deutsche Forschungsgemeinschaft.]

P27. Automatic speech recognition for speech-impaired people. Gloria Stevens Carlson, Jared Bernstein, and Donald W. Bell (Speech Research Program, SRI International, 333 Ravenswood Avenue, Menlo Park, CA 94025)

SRI has been evaluating automatic speech recognition for text input by speech-impaired people. The speech of 85 speech-impaired people (primarily deaf and cerebral palsy) was recorded. This speech was input to a high-performance, template-based speech recognizer. With a 300-word vocabulary, the machine recognized approximately 25% of the impaired speakers better than human listeners did (in context); with a 50-word vocabulary, the machine recognized approximately 57% better than listeners. A preprototype text-entry and communication aid is being evaluated in our laboratory. Twelve subjects have tried this system briefly, four of whom could use speech input faster than typing input after only a few minutes of practice. Two subjects (who were not immediately faster with speech input) were trained and evaluated on the system more extensively. Both subjects' performance with the speech recognizer improved dramatically; after 9 h of practice, one's speaking rate almost doubled his manual typing rate. This presentation will describe this system, discuss our results, and show videotapes of the two trained users. [Work supported by NIH.]

P28. Sensory integration of speech by a profoundly deaf subject using tactile aids. Michael P. Lynch, Rebecca E. Eilers, D. Kimbrough Oller, and Patricia J. Pero (Department of Psychology and Pediatrics, University of Miami, Mailman Center for Child Development, P. O. Box 016820 (D-820), Miami, FL 33101)

Previous research on tactual speech perception has focused on the relative contributions of lipreading and taction with normally hearing subjects. The integration of information from touch with that from lipreading and aided hearing by profoundly deaf subjects has not been investigated. In the present study, a profoundly deaf adult with previous tactual vocoder training [Lynch *et al.*, J. Acoust. Soc. Am. Suppl. 1 **82**, S22 (1987)] was unable to identify a group of words in any one of three conditions, lipreading (L), aided hearing (H), or touch. The tactile devices used were a multichannel electrocutaneous vocoder (TV) and a two-channel vibrotactile aid (TA). To test sensory integration, the words were randomly assigned to one of seven conditions: (a) L + H, (b) H + TV, (c) H + TA, (d) L + TV, (e) L + TA, (f) L + TV + H, and (g) L + TA + H. Results indicate that: the subject integrated information across all three modalities, yielding highest performance in the L + TV + H and L + TA + H conditions; integration occurred when both lipreading and touch were used; integration occurred when both lipreading and aided hearing were used; and each of the tactile devices provided substantial speech information. Analyses of the subject's error patterns and sentence identification performance will also be presented. [Work supported by NIDRR, Dade County Public Schools, and NIH.]

P29. Effects of number of channels on speech perception with tactile aids. Janet M. Weisenberger (Central Institute for the Deaf, 818 S. Euclid, St. Louis, MO 63110)

Although there is considerable evidence that multichannel tactile aids for the hearing impaired provide more information about the speech signal to the wearer than do single-channel tactile aids, it has not been determined how many channels are necessary for optimal transmission of speech information. In the present study, the 5-channel Tactaid V (Audiological Engineering Corporation) and the 16-channel Queen's University tactile vocoder were evaluated in a series of phoneme identification and connected speech tasks, to determine whether the additional channels present in the 16-channel design lead to better speech perception performance. Results showed that, in relatively simple tasks, such as minimal pairs phoneme discrimination, subjects performed at similar levels with both devices. However, in more complex tasks, such as integration with lipreading, the 16-channel device yielded significantly higher levels of performance. In connected discourse tracking, both devices showed significantly higher levels of performance in conjunction with lipreading than were found under lipreading alone conditions. Results are discussed with respect to the benefits provided by a tactile aid relative to the practical limitations of tactile aids that employ a large number of tactile transducers. [Work supported by NIH and NSF.]

P30. Development of speech perception and speech production of children who used vibrotactile aids. Adele Proctor (Northeastern University, Boston, MA 02115) and Moise H. Goldstein, Jr. (Johns Hopkins University, Baltimore, MD 21218)

Since 1977, an extensive video and audio tape database on the use of single-channel vibrotactile aids by prelingually deaf children has been developed. Subjects' ages ranged from 6 weeks to 5 years at the beginning of vibrotactile aid usage. To date, subjects have been followed from periods ranging from 4 to 12 years old. Formal and standardized tests of language, lipreading, speech perception, and speech production were administered as a means of assessing children's progress. Results suggest that aid usage encourages vocalization and facilitates awareness of environmental and speech sounds, learning of lipreading, and acquisition of reading and other aspects of language. Samples of video recordings that demonstrate speech perception and speech production will be shown. [Work supported by DOE, Grant No. 133 GH70189, Department of Health and Human Services Grant No. RR07143, National Foundation March of Dimes.]

P31. Time-varying/frequency-varying amplitude compression in compensating for recruitment of loudness. Janet C. Rutledge and Mark A. Clements (School of Electrical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

Research in the area of compensating for sensorineural hearing losses, including recruitment of loudness, has indicated that amplitude compression may be a solution. Previous systems developed have used filtering techniques that operate directly on the speech waveform and have parameters that remain constant with time. A system is proposed utilizing the sinusoidal speech model [J. R. Lias and M. A. Clements, J. Acoust. Soc. Am. Suppl. 1 **82**, S5 (1987)], which allows amplitude compression to vary with both time and frequency. The system uses an iterative procedure to determine the best compression coefficient for each sinusoidal component in each 10-ms frame of speech based on a model of the impaired person's masking profile. This allows the processing to adapt to the changing properties of the speech signal in addition to the frequency characteristics of the person's residual hearing. Since the operations are performed on the sinusoidal components rather than on the speech waveform, the distortions inherent in the multichannel filtering techniques have been avoided.

P32. Videophonetics. Harry Levitt (Center for Research in Speech and Hearing Sciences, City University of New York, 33 West 42nd Street, New York, NY 10036)

Techniques developed for the acoustic analysis of speech have been applied in an analogous manner to video speech signals. This new field of investigation is referred to as videophonetics. New techniques that have been developed include video speech synthesis by means of cartoons or by concatenation of prerecorded diphones and triphones, analysis of phonetic contrasts in lipreading by interleaving frames of paired videorecordings, spatial high-pass and low-pass filtering of lipreading signals using bivariate Fourier transforms, and automating (in order to improve objectivity) the method of continuous discourse tracking for evaluating communication ability using lipreading and/or a sensory aid. [Research by NINCDS.]

P33. Effects of vibrotactile aid usage on vocal development during the first six months of life. Adele Proctor, Joanne Andreassi, and Hsien-Lan Wu (Northeastern University, Boston, MA 02115)

A profoundly deaf female infant was identified at age 5 weeks old and provided with a single-channel, vibrotactile aid at age 6 weeks. Video and audio tapes recorded in the home were analyzed for phonetic and acoustic changes in vocal development and compared to vocal changes in a normal-hearing, female infant recorded under similar conditions at the same ages. The purpose of this poster session is to demonstrate phonetic and acoustic similarities and differences between noncry vocal behavior of the deaf and normal-hearing infant during the first 6 months of life. Results of phonetic transcription and measurements of fundamental frequency [F_0], duration, and amplitude will be discussed relative to tactile aid usage and Waterson's [1988] theory of prosodic development in the acquisition of speech production. [Work supported by DOE, Grant No. 133 GH70189 and Department of Health and Human Services, Grant No. RR07143.]

TUESDAY AFTERNOON, 15 NOVEMBER 1988 KOHALA/KONA ROOM, 2:00 TO 4:29 P.M.

Session Q. Structural Acoustics and Vibration I: Active Vibration Control

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Kazuto Seto, Cochairman
Department of Mechanical Engineering
National Defense Academy
1-10-20 Hashirimizu
Yokosuka, 239 Japan

Chairman's Introduction—2:00

Invited Papers

2:05

Q1. Active vibration control of flexural power flow in beams. Chris R. Fuller and Luc O. Gonidou (Department of Mechanical Engineering, Virginia Tech, Blacksburg, VA 24061)

Active control of flexural power flow in infinite and semi-infinite thin elastic beams by point force inputs is analytically studied. Various forms of terminating impedances or discontinuities positioned on the beam are also considered. The influence of system parameters such as the discontinuity impedance and effects such as bending nearfield generation on the location and number of control actuators and error sensors is investigated and discussed. The mechanisms by which control is achieved are considered. It is demonstrated that these types of boundary conditions strongly influence the choice of optimal controller format. For example, if the error signal only contains information from the propagating wave component, then only one control actuator is needed for complete attenuation of flexural power flow. Finally, preliminary results of a companion experimental investigation will be discussed and compared with the theoretical developments. [Work supported by NASA Langley Research Center.]

2:25

Q2. Active vibration control of flexible structures by means of a quasimodal control method. Kenzo Nonami (Faculty of Engineering, Chiba University, 1-33 Yayoi-cho, Chiba, 260 Japan)

This paper proposes a new quasimodal control method that can actually be realized by combining the advantages of a modal control and of an output feedback. This method consists of displacement and velocity feedback. A quasimodal control method with only velocity feedback before is proposed. But some poles were still close to the imaginary axis. This method with displacement and velocity feedback is an improvement over that method only with velocity feedback. These feedback gains are also determined based on a modal analysis. The optimal gains of the new quasimodal control method coincide with the optimal gains of an output feedback method. The quasimodal control method becomes effective for cases where the number of controlled modes is more than the number of sensors and actuators.

2:45

Q3. Active control of simple modes of structural vibration. Vasundara V. Varadan, X.-Q. Bao, J.-H. Jeng, and Vijay K. Varadan (Department of Engineering Science and Mechanics, The Pennsylvania State University, University Park, PA 16802)

An active feedback control system is developed for control of beam vibration and torsional vibration. The experimental system consists of piezoceramic sensors and actuators bonded on a 32-cm aluminum beam as well as electronic circuits incorporating a negative feedback loop. Detailed analysis of the system is presented, and the obtained data are analyzed. [Work supported by the Research Center for the Engineering of Electronic and Acoustic Materials.]

3:05

Q4. Active control of impact vibration using an active damper with a preview action. Nobuo Tanaka (Mechanical Engineering Laboratory, 1-2 Namiki, Tsukuba, 305 Japan)

In order to suppress impact vibration, an active control method using an active damper with a preview action is presented. First, from the viewpoint of the preview control method, the system equations of the preview control system are derived. Then, using a parameter optimization technique, the necessary conditions for optimality are obtained. Next, by solving the optimal conditions, the design parameters of the active damper system are determined. Based upon the design procedure, the analytical results of the control effect are shown, and the roles of both the preview control and the feedback control on the control effect are investigated. With a view to clarifying the mechanism of the vibration control with the preview action, a control chart is introduced. Using the chart, the superiority of the preview control method over other control methods is discussed. Moreover, a hybrid control compensator is presented, which consists of a digital and analog component, that is, one to control the preview time and the other to calculate the control signal, respectively. Finally, an experiment is carried out, verifying the possibility of suppressing an impact vibration.

3:25

Q5. Active vibration control for a precision pointing system on the Space Station. Samuel W. Sirlin (Jet Propulsion Laboratory, California Institute of Technology, Mail Stop 198-326, 4800 Oak Grove Drive, Pasadena, CA 91109)

Disturbance attenuation is receiving increasing attention for spacecraft as it becomes evident that precise pointing is required in disturbance-rich environments. Future observational instruments have extremely tight pointing and stability requirements (for example, 50 nanoradian stability for a proposed planet seeking Circumstellar Imaging Telescope). At the same time, these instruments may be mounted on board the Space Station, together with various machinery and people that continuously generate noise which excites the flexible station structure and tends to disturb pointing payloads. The softmounted inertially reacting pointing system (SIRPNT) combines a precise inertial pointing system, as would be used on a free-flying spacecraft, with an actively controlled (piezoelectric polymer based) softmount capable of keeping the payload tied to the basebody but suppressing the vibration that would be ordinarily transmitted across the mount. Previously presented proof-of-concept analysis has been extended to predict the system performance in a realistic environment. Optimization of the softmount control system is done to minimize the transmitted vibration subject to sensor and actuator limitations.

3:45

Q6. Noise reduction by an active vibration controller. Kazuto Seto (Department of Mechanical Engineering, National Defense Academy, 1-10-20 Hashirimizu, Yokosuka, 239 Japan)

This paper proposes a method of noise reduction by way of controlling the vibration of a wall between two rooms by an active vibration controller. This is done by using an additional sensor to measure the wall vibration and feed it back to the controller. The boardspeaker is used primarily to generate sound by exciting the board. However, in this study, it is also used as the actuator. The performance of the controller is tested on a veneer board, which is $1200 \times 900 \times 5$ mm in size, under sound vibration by another speaker. The evaluation of this method is performed experimentally by comparison of the dynamic deflection of the board with and without the controller. The experimental results show a significant reduction level to 1/10 of the board dynamic deflection at frequencies below 100 Hz. The effectiveness of the method is confirmed theoretically by analog simulation studies. Also, the stability of the controller is examined using the root locus method.

Contributed Papers

4:05

Q7. Active acoustic absorber for plane waves in water. X.-Q. Bao, Vijay K. Varadan, and Vasundara V. Varadan (Department of Engineering Science and Mechanics, The Pennsylvania State University, University Park, PA 16802)

Presented here is the design of an active acoustic absorber that absorbs normally incident plane waves and completely eliminates reflection. The absorber consists of incoming and reflected wave sensors, an absorption transducer, and an associated control system. The signal due to the incoming wave is sensed, then amplified, using an amplifier with an appropriate transfer function, and then used to actuate the absorption transducer. The error of elimination is detected by the reflected wave sensor, with the control system making the error approach a zero value. Data from experiments conducted in an acoustic pulse tube are also presented. [Work supported by the Research Center for the Engineering of Electronic and Acoustic Materials.]

4:17

Q8. Modeling and vibration control of multi-link flexible arms. Shinya Ishii, Chiaki Yasuda, and Katsuhisa Fujita (Mitsubishi Heavy Industries, Ltd., Takasago R & D Center, 2-1-1 Shinhama, Arai-cho, Takasago, 676 Japan)

Linear state-space models describing the motions of single-link and two-link flexible manipulators are derived by an assumed mode method. These motions are assumed to be a linear combination of a vibration due to flexible deflections and a rigid-body motion. By assuming the flexible arms to be Bernoulli–Euler beams, their vibration behaviors are described by using the partial differential equations with well-posed boundary conditions, which constitute eigenvalue problems. These problems can be solved either by the symbolic processing system or by the finite element method. The controller is designed by the optimal regulator control theory using the quadratic cost function. The stress at the link's root, the angular displacement of the hub, and the angular velocity of the motor are adopted as the feedback signals. The experimental systems of these single-link and two-link models are assumed to have planar motions, and the relative motions of the two links are assumed to result from torques applied at each joint of the systems. Also, the second joint and end effector are assumed to be supported by air bearings on a flat table so as to eliminate the effects of gravity. The experimental results will be presented. The method described above can also be applied to multi-link systems.

TUESDAY AFTERNOON, 15 NOVEMBER 1988 HONOLULU/KAHUKU ROOM, 2:00 TO 6:05 P.M.

Session R. Underwater Acoustics II: Time Domain Methods for Underwater Acoustics

Henrik Schmidt, Cochairman
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Fuchu, 183 Japan

Chairman's Introduction—2:00

Invited Papers

2:05

R1. Pulse modeling in ocean acoustics: Brute-force Fourier synthesis versus time-domain techniques. Finn B. Jensen (SACLANT Undersea Research Centre, 19026 La Spezia, Italy)

There are fundamentally two different approaches to the pulse modeling problem. The first is to generate pulse results by Fourier synthesis of time-harmonic solutions. This approach requires little programming effort and builds on the existing set of cw codes (normal modes, FFP, PE), which have been developed and continuously improved over the past decade or two. The second approach is to solve the problem directly in the time domain, which requires the development of an entirely new set of propagation codes. A similar effort would only pay off if there are significant computational advantages to solving the problem in the time domain. The issue of computational efficiency is addressed by solving characteristic short- and long-range propagation problems in shallow and deep water, employing both Fourier-synthesized cw codes (FFP and PE) and time-domain codes recently developed by M. B. Porter (TDFFP) and M. D. Collins (TDPE).

2:25

R2. Time-domain elastodynamic scattering problems. J. D. Achenbach (The Technological Institute, Northwestern University, Evanston, IL 60208), S. Hirose (Okayama University, Okayama, 700 Japan), and Ch. Zhang (Tongji University, Shanghai, People's Republic of China)

The interaction of an ultrasonic pulse with an inhomogeneity in an elastic solid is generally analyzed by the use of the FFT over time in combination with a numerical method for the scattering problem in the frequency domain. For pulses with a high-frequency content this approach tends to be computationally intensive. In this talk the possibilities of a direct analytical time-domain approach are explored, and then there is a discussion of numerical techniques. Results are presented for a single crack (slit and penny-shaped) and for macrocrack–microcrack configurations. Some of these results have been obtained by the finite difference method. The main emphasis in this paper is, however, on the development of a time-domain boundary integral equation method. By the use of appropriate representation integrals, a system of boundary integral equations has been obtained, which has subsequently been cast in a form that is amenable to a solution by the boundary element method in conjunction with a time-stepping technique. Particular attention has been devoted to dynamic overshoots of the stress intensity factors. Elastodynamic stress intensity factors have been computed as functions of time, and they have been compared with results of other authors.

2:45

R3. Acoustic radiation from an impulsive point source in a continuously layered fluid—An analysis based on the Cagniard method. Adrianus T. de Hoop (Laboratory of Electromagnetic Research, Faculty of Electrical Engineering, Delft University of Technology, P.O. Box 5031, 2600 GA Delft, The Netherlands)

Acoustic radiation from an impulsive point source in a continuously layered fluid with depth-varying parameters is investigated theoretically with the aid of the modified Cagniard method, that starts with a one-sided Laplace transformation with respect to time and a Fourier transformation with respect to the horizontal space coordinates. Using appropriate one-sided Green's functions, the system of transform-domain differential equations in the depth coordinate is rewritten as a system of integral equations that, for not too rapidly varying fluid properties, can be solved by iteration. The modified Cagniard method leads to space-time expressions for the relevant iterates. To show the generality of the method, the fluid is assumed to show anisotropy in its volume density of mass. The continuously refracted waves emitted by the source and the singly, continuously, reflected wave in an isotropic fluid are discussed in detail. With this method, no difficulties arise with "turning rays" as is the case in the frequency-domain analysis of the problem. [Work done as a Visiting Scientist at Schlumberger–Doll Research, Old Quarry Road, Ridgefield, CT 06877-4108.]

3:05

R4. Linear and nonlinear acoustics in the time domain: Long-range pulse propagation. B. Edward McDonald (NORDA, NSTL, MS 39529)

During recent years the nonlinear progressive wave equation (NPE) model has been developed as a nonlinear time domain counterpart of the linear frequency domain parabolic equation (PE) model [B. E. McDonald and W. A. Kuperman, J. Acoust. Soc. Am. **81**, 1406–1417 (1987)]. It was motivated by investigation of nonlinear wave evolution, so that time domain formulation was natural. With nonlinearity absent, the NPE gives an efficient algorithm for linear broadband propagation in the time domain. (No Fourier transforms between frequency and time domains are involved.) Recent simulation results from the NPE model (given in a color movie) illustrate linear and nonlinear pulse propagation through a deep ocean convergence

zone. Differences in physics between broadband linear and nonlinear propagation will be pointed out and discussed. Linear results reveal transients whose time scale as a function of range is "remembered" from the source. In corresponding nonlinear results, steepening and shock formation cause the source's time scale to be gradually "forgotten." For the parameters of the example given, nonlinear spreading behind the shock increases the transients' time scale by roughly a factor of 3 in the farfield.

3:25

R5. On synthetic aperture sonar. Masaaki Shishido (1st Sonar Engineering Department, Radio Application Division, NEC Corporation, 1-10 Nisshin-Cho, Fuchu, 183 Japan)

Synthetic aperture technology is used in airborne radar or on satellites like SEASAT to obtain high-resolution pictures of the Earth's surface. On the other hand, in the field of underwater acoustics, the effectiveness of synthetic aperture side-looking sonar has long been predicted. However, this has not come to practical use, mainly because of the influence of the trembling motion of the sonar platforms. First, the principle of synthetic aperture sonar and the similarity between CDP stacking of seismic exploration and synthetic aperture processing are discussed. Then, the influence of platform trembling on the synthetic aperture beamforming is quantitatively studied, and the degradation of the picture is evaluated with computer simulation to show the allowable range for the trembling motion amplitude and period against the used signal frequency.

3:45

R6. Reduction of bit-error rates by adaptive equalization for a 500-kbit/s underwater acoustic communication system. Shinji Yauchi and Akio Kaya (Systems Laboratory, Oki Electric Industry Company, Ltd., 4-11-22 Shibaura, Minato-ku, Tokyo, 108 Japan)

For the video transmission from untethered subsea robots, an acoustic communication system using the 16-ary quadrature amplitude modulation (16QAM) method is being developed to perform at a capacity of 500 kbit/s at a maximum transmission range of 60 m. To improve the quality of the transmitted information, an adaptive equalization technique will be introduced to this system. In this study, an adaptive equalizer based on a baseband decision-directed equalization method is realized as a computer program. The optimal parameters of the equalizer such as the number of taps and step sizes are determined by computer simulations. In order to estimate the performance of the acoustic communication system with an adaptive equalizer, random code data modulated by 16QAM are transmitted to a receiver located 10–60 m from the source through a surface channel, and stored in a digital memory bank for the off-line processing of an adaptive equalizer. Results show that the bit-error rates in the output of the receiver are about 10^{-4} at a range of 60 m because of the channel distortion caused by multipath interference, the attenuation characteristics of the channel, etc. However, the bit-error rates in the output of the adaptive equalizer are reduced to less than 10^{-7} in the 10- to-60-m range. [Conducted as part of the R&D Program of the Large-Scale Project "Advanced Robot Technology" by the Agency of Industrial Science and Technology, Ministry of International Trade and Industry.]

Contributed Papers

4:05

R7. Wave field factorization for broadband acoustic signals. John J. McCoy (Department of Civil Engineering, CUA, Pangborn Hall G-10, Washington, DC 20064)

A framework has been demonstrated for factoring the Helmholtz equation, governing for narrow-band acoustic signals, into a pair of one-way equations [L. Fishman and J. J. McCoy, J. Math. Phys. **25**, 285–296 (1984)]. This factoring requires a range-independent propagation environment. The factored Helmholtz equation has been demonstrated to provide a marching algorithm [L. Fishman and J. J. McCoy, Geophys. J. R. Astron. Soc. **80**, 439–461 (1985)], which is a generalization of the split-step algorithm, the latter being frequently used for marching the solution of the ordinary parabolic wave equation. The extension of these results to broadband acoustic signals is considered. Two factorizations are considered and shown to apply to different initial/boundary value problems. The first factorization is based on a Fourier synthesis of that obtained for narrow-band signals. This applies for a forcing problem as a broadband time series acting across a source range plane. The second factorization obtains for the time coordinate of the wave equation. This applies for an infinite spatial domain problem for specified initial time data.

4:17

R8. The finite difference method for time domain solutions to range-dependent bottom-interaction problems. Ralph A. Stephen (Woods Hole Oceanographic Institution, Woods Hole, MA 02543)

An explicit second-order finite difference scheme has been used to solve the elastic wave equation in the time domain. Solutions are presented for the perfect wedge, the lossless penetrable wedge, and the plane parallel waveguide, which have been proposed as benchmarks by the Acoustical Society of America. Good agreement with reference solutions is obtained if the media is discretized at 20 grid points per wavelength. The principle disadvantage of the technique is long computational times that are between 10 and 20 h on a minicomputer without an array processor. The method has the advantage of providing phase information and, when run for a pulse source, of providing insight into the evolution of the wave field and energy partitioning. Arbitrarily more complex models including velocity gradients, strong lateral heterogeneities, and random media can be solved with no additional computational effort. The method has also been formulated to include shear wave effects. [Work supported by ONR.]

4:29

R9. Time domain finite difference modeling of acoustic wave scattering from an elastic cylinder. Martin E. Dougherty (Massachusetts Institute of Technology/Woods Hole Oceanographic Institution Joint Program, Woods Hole, MA 02543) and Ralph A. Stephen (Woods Hole Oceanographic Institution, Woods Hole, MA 02543)

A 2-D finite difference formulation of the elastic wave equation has been used to investigate seismo/acoustic scattering from laterally heterogeneous oceanic crust. In an effort to validate the use of this code to handle such problems, the same formulation has been used to solve the canonical problem of scattering of an acoustic wave from an infinite elastic circular cylinder. Analytical theory predicts that an acoustic wave will be scattered only into certain distinct azimuths after interaction with a cylinder of given elastic parameters. These scattering angles and amplitudes are very sensitive to both the frequency of the source and the diameter of the cylinder. The finite difference results of scattering of both a relatively broadband plane wave pulse and a series of monochromatic cw plane waves at different frequencies are presented. In general, the scattering pattern produced by the pulse source does not agree well with the analytical solution for the pulse center frequency. This is due to the relatively broad frequency band of the pulse. However, scattering patterns and amplitudes produced by the monochromatic cw plane wave sources do agree very well with the expected analytical solutions.

4:41

R10. Inversion of wave field data by simulated annealing. Atanu Basu and L. Neil Frazer (HIG, University of Hawaii at Manoa, Honolulu, HI 96822)

Simulated annealing is a new Monte-Carlo optimization method that (if properly used) does not get stuck in local minima of the objective function. This feature is attractive for the inversion of wave field data where the lack of low-frequency energy makes cycle skipping a problem. To gain experience with the method, it was applied to a simple problem: determining a sound-speed profile $c(z)$ from recordings of two shots at unequal distances from a vertical array. For a nearly stratified ocean this problem is equivalent to a single shot, recorded on two vertical arrays. A profile $c(z)$, propagating data from the near array to the far array, is assumed and then the propagated data are compared with the actual data at the second array. If the propagated and actual data agree then $c(z)$ is a good profile. If the sound-speed profile has, say, 15 parameters with, say, 10 possible values for each parameter then the number of possible profiles is 10^{15}. In tests with synthetic data, simulated annealing gave a nearly correct profile with a few hours of iteration (on a desk-side workstation). Experiments are currently going on with different annealing schedules. Results so far suggest that critical temperature (T_{cr}) is difficult to determine, and so annealing strategies which do not require a prior estimate of T_{cr} are more efficient than those which do. [Work supported by ONR.]

4:53

R11. Numerical experiments with time-domain localization. Peter I. Pecholcs and L. Neil Frazer (HIG, University of Hawaii at Manoa, Honolulu, HI 96822)

Time-domain localization is of interest because: (a) ambiguity surfaces constructed using frequency-domain methods have large sidelobes unless many frequencies are used; (b) time-domain Green's functions can be rapidly constructed using ray theory; and (c) in theory, time-domain methods can be used to directly determine the velocity, as well as the position, of the source. Here some natural time-domain algorithms are applied to a homogeneous waveguide with a soft top and hard bottom. Clay's matched filter approach, and a technique known in the oil industry as reverse time migration, are attractive because neither requires a knowledge of the source waveform. If the source waveform is available (even though its time origin is not) then one can localize using a Bartlett, or a maximum likelihood, formula directly in the time domain. A concurrent search is needed to determine the best time origin for the source waveform. If the source waveform is available, then one can localize in velocity space by use of a ray theoretical Green's function in which the source waveforms of different rays are compressed or expanded according to the component of velocity parallel to each ray at a candidate source location. [Work supported by ONR.]

5:05

R12. The time-domain parabolic wave equation and its path integral representation. Dalcio K. Dacol (Acoustics Division, Code 5160, U.S. Naval Research Laboratory, Washington, DC 20375-5000)

A path integral representation for the solution of a time-domain parabolic wave equation (TDPE) was developed. It shows promise for describing propagation of sound pulses in the ocean when boundary interactions can be neglected. The TDPE is a linearized version of McDonald and Kuperman's nonlinear progressive wave equation [B. E. McDonald and W. A. Kuperman, J. Acoust. Soc. Am. **81**, 1406 (1987)]. Using the techniques developed by these authors one can derive the TDPE from the acoustic wave equation thus showing that the TDPE's solution is an approximate solution to the acoustic wave equation [M. D. Collins, J. Acoust. Soc. Am. Suppl. **82**, S122 (1987)]. The TDPE has the mathematical structure of a time-dependent Schrödinger equation and a phase space path integral representation for its solution can be constructed by well-known methods. Applications to be discussed include propagation in a sound channel and in randomly fluctuating media. [Work supported by U.S. Naval Research Laboratory.]

5:17

R13. Numerical modeling of acoustic emission from propagating cracks in an Arctic ice cover. Henrik Schmidt and Jae Soo Kim (Massachusetts Institute of Technology, Cambridge, MA 02139)

An earlier developed numerical model for three-dimensional propagation in horizontally stratified fluid/elastic media [H. Schmidt and J. Glattetre, J. Acoust. Soc. Am. 78, 2105–2114 (1985)] has been modified to incorporate seismic moment representations for compact cracks of tensile, dip-slip, and strike-slip types. The applied global matrix approach first yields the depth-dependent Green's function simultaneously in all layers for an arbitrary number of individual source depths. The frequency domain solution for a three-dimensionally propagating crack is then determined by wavenumber integration over a spatial grid of superpositions of individual source contributions multiplied by the proper phase terms. Finally, the time domain solution over the spatial grid is obtained by simple Fourier synthesis. The present Fourier transform approach has the advantage that many crack propagation scenarios can be treated with only a single computer intensive solution of the wave equation. Once the depth-dependent Green's function has been determined for the relevant crack type and depths, the field can be evaluated very efficiently by superposition for any crack propagation direction and speed. The developed model has been applied to theoretical analysis of the sound field produced in an Arctic environment by ice cracking, demonstrating the effect of cracking mechanisms, propagation speed as well as environmental parameters in general. [Work supported by ONR.]

5:29

R14. Simulated autocorrelations of a broadband signal in an ocean with a layered bottom. Gordon R. Ebbeson (Defence Research Establishment Pacific, FMO, Victoria, British Columbia V0S 1B0, Canada)

It is well documented that the correlations between the multipath arrivals from a broadband acoustic source in a deep ocean can be greatly affected by the bottom sediment. As an aid to explaining these effects, the seismogram version of the SAFARI model was extended to simulate the autocorrelation of the received signal. As with the original model, the bottom environment is input in the form of a geoacoustic bottom model. However, the inputs to the extended model also include the source level, realistic ambient noise and sea state conditions, noise due to distant shipping, and the self-noise levels of the receiving array. Simulations were carried out using the environment of the Tufts Abyssal Plain of the Northeastern Pacific. The geoacoustic bottom model for this area was developed at DREP [N. R. Chapman, J. Acoust. Soc. Am. 73, 1601–1607 (1983)] and has been validated by a detailed analysis of propagation loss measurements. It was found from these simulations that reflections from both the water–sediment and sediment basement interfaces result in a time-spreading of the correlogram peaks as well as a reduction in the overall correlation magnitude.

5:41

R15. The effects of time delay quantization on underwater ultrasonic imaging techniques. A. H. Goode, J. M. Reeves, and S. O. Harrold (Department of Electronic Engineering, Portsmouth Polytechnic, Portsmouth, Hampshire PO1 3DJ, United Kingdom)

This paper describes different pulse echo imaging techniques based on time delay focused arrays for both transmission and reception. The effects on system performance of time delay quantization are examined using a versatile pulse echo imaging test facility and a corresponding computer simulation. The system currently used has B and C scan capability over a 40° sector. It is designed around 32-element linear-phased arrays, operating at 2 MHz, for both transmission and reception of short bursts of ultrasonic energy. The techniques described fall into two categories, high resolution and low resolution. The high-resolution techniques calculate the sum of the corresponding instantaneous samples of each of the 32 individually received signals, therefore allowing phase cancellation/reinforcement to occur. The low-resolution technique simply utilizes the total energy in the received signal. A brief description of the test facility is included together with the methods for using these types of imaging techniques. The sensitivity effects due to the time delay quantization are discussed for various levels of quantization, and three-dimensional sensitivity contour plots depicting system performance are included.

5:53

R16. An application of the broken mirror approximation to modeling bistatic reverberation in the ocean. Henry Weinberg (SYNTEK Engineering & Computer Systems, Inc., 1 Denison Avenue, Suite 202, Mystic, CT 06355-2709)

The Navy Interim Surface Ship Model (NISSM II), a monostatic reverberation model, was applied to data measured by the Canadian Defence Research Establishment Pacific (DREP). NISSM II could not explain the behavior of certain discrete arrivals. The source of the discrepancy was eventually attributed to the bistatic nature of the experiment. In 1986, the Generic Bistatic Reverberation Model was developed and applied to the data without much success. Other modelers were already aware that direct transmitter–receiver arrivals, the counterpart of vertical fathometer returns in the monostatic case, could be significant in bistatic reverberation measurements. When these direct arrivals were added, it became obvious that the source of discrepancy between the model and the data had been isolated, but not resolved. The implementation of a "broken mirror" model provided the missing link for explaining the DREP data. This paper highlights the results of the bistatic investigation.

TUESDAY AFTERNOON, 15 NOVEMBER 1988 — MOLOKAI ROOM, 3:00 TO 5:00 P.M.

Session S. Physiological Acoustics III: Workings of the Cochlea and Auditory Nervous System (Poster Session)

Harunori Ohmori, Cochairman
National Institute for Physiological Science
Myodaiji-cho
Okazaki, 444 Japan

Mario A. Ruggero, Cochairman
Department of Otolaryngology
University of Minnesota
Minneapolis, Minnesota 55414

Contributed Papers

Posters must be set up before 2:00 p.m. (before start of Session O). All posters will be displayed from 3:00 to 5:00 p.m. To allow contributors the opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:00 to 4:00 p.m. and contributors of even-numbered papers will be at their posters from 4:00 to 5:00 p.m. Posters must be removed promptly at 5:00 p.m. to prepare the room for the buffet social.

S1. Biochemical considerations in presbycusis. Isolde Thalmann, Kuniaki Takahashi, and Ruediger Thalmann (Department of Otolaryngology, Washington University, St. Louis, MO 63110)

Virtually no direct information about biochemical changes in the peripheral auditory system due to aging is available. However, numerous predictions about biochemical changes due to characteristic degeneration patterns can be made indirectly. (1) Since glycogen and calmodulin are extremely high in outer hair cells, significant degeneration of these cells due to aging will result in lowered average levels of both substances in whole organ of Corti. Analogous considerations apply if major longitudinal gradients are present (e.g., glycogen, *P*-creatine, cyclic GMP). (2) Degeneration of stria vascularis, the so-called "metabolic" type of presbycusis, will result in a lowered endolymphatic potential and low endolymph K, while Na and Ca will be elevated. (3) The "inner ear conduction" type of presbycusis is associated with massive degeneration of the spiral ligament (SL). Apart from mechanical effects on basilar membrane motion, this should result in significant changes in perilymph chemistry. The normal SL exhibits extremely high carbonic anhydrase, which is thought to be essential in perilymph regulation. The SL also appears to be responsible for the markedly different kinetics of K and furosemide in scala vestibuli and tympani. The pronounced gradients of these substances between the scalae can no longer be maintained in the absence of a functional SL.

S2. Electrochemical aspects of calcium ions in the stria vascularis. Katsuhisa Ikeda and Tetsuo Morizono (Department of Otolaryngology, Research East, 2630 University Avenue S.E., Minneapolis, MN 55414)

The dc potential and the Ca concentration of the stria vascularis were measured using double-barreled Ca-selective microelectrodes with fine tips. Healthy chinchillas were anesthetized with ketamine hydrochloride and artificial ventilation was maintained under conditions of muscular relaxation. The microelectrode was advanced through the spiral ligament by an autodrive manipulator. The dc potential from the spiral ligament and the Ca concentration (mean $\pm$ s.d.) are respectively as follows: 0 mV and 1.7 ± 0.3 mM in the spiral ligament ($n = 11$); 6.3 ± 7.8 mV and $2.4 \pm 2.3\,\mu$M in basal cells ($n = 10$); 55.2 ± 5.8 mV and 0.68 ± 0.20 mM in the intrastrial space ($n = 5$); 72.9 ± 9.8 mV and $0.38 \pm 0.28\,\mu$M in marginal cells ($n = 11$); 78.3 ± 2.3 mV and $17 \pm 7\,\mu$M in endolymph ($n = 11$). The transport mechanism of calcium ions in relation to the localization of Ca-ATPase is discussed. [Work supported by NINCDS.]

S3. Effects of scalae tapering, perilymph viscosity, helicotrema, and the cochlear map function (CMF) on models of the cochlea. Sunil Puria and Jont B. Allen (AT&T Bell Laboratories, 600 Mountain Avenue, Room 2D-504, Murray Hill, NJ 07974)

The basilar membrane response was calculated by cascading two-port ABCD matrices that are formulated to include effects due to scalae tapering, perilymph viscosity, and helicotrema. Previously, Koshigoe *et al.* [J. Acoust. Soc. Am. **74**, 486–492 (1983)] showed that perilymph viscosity has an important effect on cochlear input impedance (Z_c) in the frequency region below 500 Hz. It is shown that Z_c is affected more by scala tapering than by perilymph viscosity, and it is affected very little by the helicotrema. It is also shown that low-frequency standing wave (LFSW)'s seen in Z_c of several models [J. W. Mathews, Ph.D. thesis, Washington University, Missouri (1980)] are strongly dependent on the CMF used. LFSW's present in nonlinear time domain solutions are likely to manifest themselves across the entire frequency spectrum and as a result mislead one's conclusions regarding basilar membrane mechanics. Previous models have used a straight CMF of the form $K_0 \exp(-ax)$ resulting in LFSW's. Liberman's CMF leads to a reduction of LFSW's and when Greenwood's drooped CMF [J. Acoust. Soc. Am. **33**, 1344–1356 (1961)] is used LFSW's are not present.

S4. A local cochlear mechanism of endolymph volume regulation. Ruediger Thalmann, Alec N. Salt, and John DeMott (Department of Otolaryngology, Washington University, St. Louis, MO 63110)

It is generally believed that endolymph volume is regulated, not by the cochlea, but by the endolymphatic sac. Acute changes of endolymph volume in anesthetized guinea pigs has been induced by perfusing hypertonic medium (400 mOsm/kg H_2O) for 20 min through the perilymphatic spaces. During hypertonic perfusion, changes of endocochlear potential (EP), endolymph K level, and endolymph volume (using tetramethylammonium : TMA as a volume marker) were monitored using ion-selective microelectrodes. The osmotically induced K increase appeared to be comprised of two components. The first, an increase due to the loss of water from endolymph was estimated from the recorded TMA changes. In addition, a second component existed representing a net addition of K to the endolymph. This component was closely correlated with the EP reduction during dehydration, which suggests the endolymph K increase may result from the admission of anions into endolymph. A mechanism that produces an increase in endolymph solutes when endolymph volume is reduced would allow volume changes induced by osmotic treatment to be counteracted. These data suggest that, in addition to the ionic composition of endolymph being locally regulated, endolymph volume may also be locally controlled. [Work supported by NIH and DRF.]

S5. Apical hair cells and hearing. David B. Moody, Cynthia A. Prosen, Mitchell S. Sommers, William C. Stebbins (Kresge Hearing Research Institute, University of Michigan Medical School, Ann Arbor, MI 48109), and David W. Smith (Department of Otolaryngology, University of Toronto, Toronto, Ontario M5S 1A8, Canada)

Hair cells located in the cochlear apex were destroyed by low-frequency noise exposure or by application of a -140 °C cryoprobe to the bony wall of the apical turn. Complementary lesions that *spared* apical hair cells were produced by treatment with ototoxic antibiotics. Behavioral thresholds were measured in monkeys, chinchillas, and guinea pigs before and after these treatments. The results showed (1) with loss of up to 80% of apical outer hair cells, thresholds were unchanged; (2) with complete loss of both outer and inner apical hair cells, thresholds shifted approximately 20 dB at low frequencies; and (3) with complete loss of hair cells *except* in the apex, some hearing remained, with better hearing at low frequencies. Addition of a high-pass masker always elevated high-frequency thresholds, but only elevated low-frequency thresholds when all apical hair cells had been destroyed, indicating that basal hair cells can respond to low-frequency stimuli. Compared to data from more basal portions of the cochlea, these data suggest that functional differences exist between the base and apex, with comparable damage in the apex resulting in less hearing impairment. [Work supported by NINCDS grants NS05785 and NS25564.]

S6. Development of auditory function and cochlear morphology in the rat. L. P. Rybak, C. Whitworth, and V. Scott (Department of Surgery, SIU School of Medicine, P.O. Box 19230, Springfield, IL 62794-9230)

Auditory function was studied in the rat pup using cochlear potentials [endocochlear potential (EP) and compound action potential (CAP)] and auditory brainstem response (ABR). EP development followed a sigmoid curve. CAP and ABR threshold responses to clicks were detected at about 13 days of age and showed a progressive decrease in threshold until adultlike levels were attained at 30 days. Parallel studies of organ of Corti structural development were carried out using scanning electron microscopy (SEM). Interesting changes in the width of the organ of Corti and stereocilia configuration were observed. Correlation of onset of detectable auditory threshold and morphologic changes will be presented. [Work supported by DRF.]

S7. Human hair cell–VIII nerve transmission deficit: A case study. Arnold Starr, David McPherson, Julie Patterson, and Cam Walker (Department of Neurology, University of California Irvine, Irvine, CA 92717)

A 12-year-old girl with bilateral moderate pure tone hearing loss (30–50 dB) and profound speech comprehension impairment (0%–30% correct discrimination) was tested with auditory-evoked potentials. *No* neural components occurred in the brainstem (1–10 ms), mid (10–50 ms), and long-latency (50–250 ms) time domains even though the patient could "hear" the clicks. A cognitive P300 component was evoked during the discrimination of an infrequent tonal signal indicating normal stimulus classification function. Cochlear microphonic activity of normal amplitude was recorded from an ear-canal electrode. Thus, the absence of neural components of auditory-evoked potentials in the presence of preserved receptor (hair cell) function is interpreted as due to an alteration of transmission between hair cell and VIII nerve leading to a loss of synchrony of VIII nerve activity and an absence of evoked potentials. The finding that the patient's temporal discrimination abilities were severely impaired supports such an interpretation: binaural time or intensity cues could not be used for localization; binaural "beats" were absent; monaural resolution of two clicks was markedly slowed.

S8. Spectral characteristics of the responses of primary auditory-nerve fibers to frequency-modulated signals. S. M. Khanna and M. C. Teich (Departments of Electrical Engineering and Otolaryngology, Columbia University, New York, NY 10027)

The spectral responses of cat single primary auditory nerve fibers to sinusoidal frequency-modulated (FM) acoustic signals applied to the ear are reported. Period histograms were constructed from the neural spike-train data, and the frequency spectrum was determined by Fourier transforming these histograms. Several clusters of spectral components were present. The lowest-frequency cluster consists of components at dc, at the modulation frequency, and at its harmonics. In the next cluster, components surround the carrier frequency and are separated from it by the modulation frequency and its harmonics. Higher-frequency clusters surround frequencies that are twice and three times the carrier frequency. The components in each cluster are separated from the multiples of the carrier frequency by the modulation frequency and its harmonics. The magnitudes of the spectral components were investigated for carrier frequencies located below, at, and above the unit characteristic frequency (CF), and for different signal levels, modulation frequencies, and modulation indices. The components at the modulation frequency and its harmonics were strong and present over a wide range of signal levels, carrier frequencies, modulation frequencies, and nerve-fiber characteristics. The presence of components at the modulation frequency indicates that a demodulation process is occurring. This process may be significant for speech recognition. [Work supported by NIH and NSF.]

S9. Fractal character of the auditory neural spike train. M. C. Teich (Department of Electrical Engineering, Columbia University, New York, NY 10027)

Long-counting-time pulse-number distributions (PNDs) were measured in cat primary auditory fibers using a broad range of fibers, stimuli, counting times T, and number of repetitions N_T. Short-counting-time PNDs readily exhibit the existence of spike pairs, whereas PNDs with $T \gtrsim 100$ ms exhibit irregular shapes in all cases, indicating the presence of spike clusters in the underlying auditory neural spike train. The count variance-to-mean ratio (or Fano factor) $F(T)$ was relatively constant over a broad range of stimulus levels for all units measured, but increased substantially as T or N_T increased. A normalized coincidence rate [$g(\tau)$ versus delay time τ], based on three physiologically plausible factors, is in accord with the Fano-factor versus counting-time function. The observed power-law growth for the Fano factor for large T [$F(T) \propto T^{\alpha}$] implies that $g(\tau) \propto \tau^{\alpha-1}$ and that the spectral density of the spike train $S(f) \propto f^{-\alpha}$, where $0 \leqslant \alpha \leqslant 1$. This suggests that the auditory events exhibit fractal behavior for sufficiently large times (sufficiently low frequencies) [M. C. Teich, IEEE Trans. Biomed. Eng., in press (1989)]. This behavior leads to spike clusters in the PND. The fractal dimension for the process is $D = \alpha \approx \frac{1}{2}$ in the range $0.1\ \text{s} \leqslant T \leqslant 10$ s. The firings of vestibular fibers do not exhibit fractal behavior, suggesting that the form of the auditory neural-firing pattern serves to effectively sample fractal natural noises. [Work supported by NIH and NSF.]

S10. Contribution of auditory evoked potentials to binaural processing. David L. McPherson, Charles Pinkston, and Arnold Starr (Department of Neurology, University of California Irvine, Irvine, CA 92717)

The contribution of the early (ABR), middle (MLR), and late (LLR) auditory evoked potentials (AEPs) to binaural interaction (BI) was studied in 17 normal young adults. AEPs were recorded by an electrode at C_z and referenced to a noncephalic electrode at C_{vii}. Filtering was between 10 and 3000 Hz for both the ABR and MLR, and between 1 and 500 Hz for the LLR. Rarefaction clicks with an intensity of 60 dB above threshold for wave V of the ABR were presented at 11.1/s for both the ABR and MLR, and at 1.7/s for the LLR. Two samples of 2000 trials each were averaged for right monaural, left monaural, and binaural stimulus presentation. The binaural interaction component (BIC) was obtained by subtracting the sum of the monaural responses from the binaural response. Maximum BI in the ABR occurred at 7.35 ms with a mean BIC of 23%. Maximum BI occurred in the MLR at 39.6 ms with a mean BI of 49%. Maximum BI of the LLR occurred at 145 ms with a mean BI of 46%. This would indicate that BI is primarily medicated at the time of the generation of the N_{40} component of the MLR and that binaural processes constitute a major function of the auditory pathway rostral to the pons. [Work supported by NIH Grant #NS-11876.]

S11. Comparison of rapid and short-term adaptation in the auditory nerve. R. L. Smith (Institute of Sensory Research and Department of Bioengineering, Syracuse University, Syracuse, NY 13244) and L. A. Westerman (Quantitative Technology Corporation, 8700 SW Creekside Place, Suite D, Beaverton, OR 97005)

Events during the first few milliseconds of auditory-nerve adaptation appear to differ qualitatively from those occurring later on. This has previously been revealed using PST analysis with various bin-width sizes, and also by dissecting the perstimulatory response into two components: rapid and short-term adaptation. In order to further evaluate the extent of these differences, a series of perstimulatory and post-stimulatory comparisons were made involving the recovery from prior stimulation and the effects of sound frequency and intensity. Post-stimulatory effects on both rapid and short-term firing rates appear to be proportional to the magnitude of the conditioning or adapting response, and otherwise to be independent of sound frequency. Following stimulation, the rapid component recovers more quickly than does the short-term component. Changes in sound frequency do not alter the amount of perstimulatory short-term adaptation, but effect the rapidly adapting component in a complex manner. This may be due in part to frequency splatter produced by non-CF tones, and also to phase locking to individual cycles of low-frequency tones. Implications for models of adaptation will be discussed. [Work supported by NSF and NIH.]

S12. Neuroanatomical basis for cochlear–cochlear interaction. Glenn C. Thompson and Ann M. Thompson (Department of Otorhinolaryngology & Communication Science, Baylor College of Medicine, One Baylor Plaza, Houston, TX 77030)

Using neuroanatomical tract-tracing methods in guinea pigs, four separate pathways were observed that connect one cochlea with either the other cochlea or with itself all via brainstem olivocochlear neurons. Since each of these pathways involved a two-neuron circuit including the posterior ventral cochlear nucleus and its projection to either ipsilateral or contralateral olivocochlear neurons, the mediating structure exists for feedback control (to the same cochlea) or feedforward control (to the opposite cochlea) with only 3-ms delay. With this short latency, these pathways have the potential to modulate the sensitivity of either cochlea after the initial onset of sound, or to modulate the neural cues resulting from differences in interaural intensity and thus may be important in localizing long duration sounds. Whatever the function, these pathways constitute the most direct connection between the cochleas thus described. [Work supported by NIH and DRF.]

S13. Time-frequency receptive fields for neurons of the cochlear nuclei. Ben M. Clopton and Patricia M. Backoff (Kresge Hearing Research Institute, 1301 E. Ann, Ann Arbor, MI 48109)

Periodic segments of wideband noise, digitally synthesized, evoked highly replicable patterns in period histograms for spike activity sampled in all subdivisions of the cochlear nuclei of guinea pig. These stimuli were transformed to time-frequency surfaces of energy density, and reverse correlation between the stimulus surface and response histogram used to estimate a weighting surface, relative to the time of spike occurrence, representing a time-frequency receptive field for the unit. A region or regions of elevated energy densities on the receptive-field surface, restricted in time and frequency before spike occurrence, were observed for all units driven by the stimuli and can be interpreted as representing excitatory afferents to the unit. For some units a region or regions of energy density lower than expected were present, presumably due to inhibitory afferents. It is possible, for most units, to predict responses to tone bursts and more complex sounds from these receptive-field surfaces. This approach, an extension of the work of de Boer, Johannesma, and others [Eggermont *et al.*, Q. Rev. Biophys. **16**, 341–414 (1983)], does not depend on phaselocking and provides a mathematical base for studying neuronal networks in these nuclei. [Work supported by the NIH and NSF.]

S14. Organization of ferret auditory cortex. James W. Fleshman and Shihab A. Shamma (Systems Research Center and Department of Electrical Engineering, University of Maryland, College Park, MD 20742)

Tone-responsive neurons were studied in the temporal cortex of pentobarbital-anesthetized ferrets. In agreement with Kelly *et al.* [Hear. Res. **24**, 111–115 (1986)], we found a region contained within the curve of the suprasylvian sulcus (SSS) in which low frequencies were represented ventrolaterally and high frequencies dorsomedially. This area, provisionally identified as AI, contains cells that are narrowly tuned and respond at short latency (15–25 ms) to tones. AI is almost completely surrounded by a belt of tone-responsive cortex with differing properties. Rostral to AI, near the anterior limb of the SSS, two regions were identified, both vigorously responsive to tones, with minimum latencies comparable to those in AI: one was characterized by broad tuning; the other displayed tonotopically ordered, primarylike tuning, perhaps corresponding to area A in the cat. Two other regions were characterized by rapidly adapting, broadly tuned responses with relatively long latencies (30–150 ms). A caudal strip extends into the posterior limb of the SSS and may exhibit a tonotopic order. Ventrolaterally there is an area with variable, long latency responses and no apparent frequency axis.

S15. Tonotopic representation of the auditory cortex of the rat. Junsei Horikawa, Susumu Ito, Yutaka Hosokawa, and Keiichi Murata (Department of Neurophysiology, Medical Research Institute, Tokyo Medical and Dental University, Chiyoda-ku, Tokyo, 101 Japan)

Representation of the best frequencies (BFs) was studied by a tungsten microelectrode method in the auditory cortex of Wistar-strain rats (200–350 g) anesthetized with Nembutal (40 mg/kg of initial and 5 mg/kg/h of supplement doses). BFs of 0.5–63 kHz were represented in the auditory cortex concentrically, with high BFs in the center and low BFs in the periphery. From the tonotopic representation, this area can be separated into two areas: the primary auditory area (AI) (posterior half), and the anterior area (A) (anterior half). However, the response latency distributions in these areas (6–14 ms were not significantly different. Other smaller areas with concentrations of high BFs were located anteroventrally (AV) and posterodorsally (PD) to the main area (AI and A). The response latencies of the neurons in AV and PD (12–24 ms) were significantly longer than those in the main area. Sixty-seven percent of the 33 tuning curves obtained from the neurons in the main area were closed (showing nonmonotonic intensity functions); 33% were open; 55% were multi-peaked; and 45% were single-peaked.

S16. Frequency dependency on the I–V interval using high-pass noise-masking derived ABR's. C. W. Ponton, J. J. Eggermont (Department of Psychology, University of Calgary, 2500 University Drive NW, Calgary, Alberta T2N 1N4, Canada), S. G. Coupland (Department of Pediatrics, University of Calgary, Calgary, Alberta T2N 1N4, Canada), and M. Dom (House Ear Institute, Los Angeles, CA)

In normal-hearing adults, there is little variation in the I–V interval as a function of stimulus intensity or the rate of stimulus presentation for the standard click-evoked auditory brainstem response (ABR). However, in a population of infants, Teas *et al.* [Hear. Res. 7, 19–54 (1982)] found that the I–V interval decreased as the center frequency (CF) of the filtered click decreased down to 1 kHz CF. The present investigation evaluated the duration of the I–V interval for frequency specific ABR's using the narrow-band responses generated by the high-pass, noise-masking procedure [Don and Eggermont, J. Acoust. Soc. Am. **63**, 1084–1092 (1978)]. In an adult population, it was found that the I–V interval was constant for all derived bands down to 1.4 kHz CF. However, a significant increase in the I–V interval was found at both 0.7 and 0.2 kHz CF. Although an explanation for the I–V interval increase is not obvious, the possibility exists that a central masking mechanism similar to that suggested by Burkard and Hecox [J. Acoust. Soc. Am. **81**, 1050–1063

(1987)] may account for the findings of this study. [Work supported by the Alberta Heritage Foundation for Medical Research (AHFMR).]

interval left ear [ge] boundary exceeded right by approximately 50 ms). Episodes can occur without changes in response time or EEG activation and thus likely arise through increased infield inhibition.

S17. Inhibition and altered cortical processing. D. M. Daly and J. A. Wada (Box 210855, Dallas, TX 75211 and University of British Columbia, Vancouver, British Columbia V6T 1S9, Canada)

When balance of inhibitions declines/fails, patients with seizures in auditory cortex report certain synthetic stops as nasals, "bleats," and then undifferentiated buzzes. Effects of increased inhibition can appear with auditory cortex in volume surrounding disinhibited areas (even contralaterally): patients report same range of stimuli as clearly identical [Daly *et al.*, J. Neurophysiol. **140**(2), 141–162 (1972)]. A case is reported with seizures in anterior temporal cortex (bilateral loci; spikes/sharp waves) and post-ictally impaired comprehension with normal peripheral acuity. Protocol included 9 h of simultaneous EEG/video monitoring and auditory testing with prerecorded sets of [be]–[de]–[ge] and [ge]–[ye]. Patient's ability to respond appropriately remained remarkably consistent during localized, bilateral, and even generalized discharge in EEG. Instances of no-response occurred when stimulus was embedded in bilateral/generalized discharge (patient once noted "missing" the stimulus). Patient also experienced episodes (20–200 s) when [de] bounds for [be]–[de] and [de]–[ge] increased approximately 200 Hz, or when [ge] bound for [ge]–[ye] increased more than 30 ms (during one 15-min

S18. Functional-activation asymmetries in normal humans studied with quantitative EEG (QEEG): First tests in the CNS Project. Judith L. Lauter (University of Arizona, Tucson, AZ 85721)

Work with behaviorally defined asymmetries such as relative ear advantages has led to the CNS Project, designed to apply noninvasive techniques such as psychophysics, EPs, QEEG, MEG, MRI, and PET to study human brain responses during functional activation. Preliminary results of a QEEG test series involving both auditory and hand-movement conditions indicate that QEEG power asymmetry patterns reflect at least two types of asymmetry organization: (1) "side of space," e.g., right-hand movement elicits a power asymmetry favoring the left hemisphere, and v.v; and (2) asymmetries based on "higher-level" principles of organization, e.g., coordination during bilateral hand movement, or differential activation based on the physical characteristics of test sounds. As with behavioral patterns of relative ear advantages, QEEG shows individual differences in detail but group agreements in overall patterns of response. It is believed that this is the first report of QEEG used for studying functional activation in healthy human subjects, and illustrates its potential usefulness for studying human neurophysiology. [Work supported in part by AFOSR.]

TUESDAY AFTERNOON, 15 NOVEMBER 1988 KOHALA/KONA ROOM, 4:35 TO 5:59 P.M.

Session T. Structural Acoustics and Vibration II: Structural Radiation

Mauro Pierucci, Chairman

Department of Aerospace Engineering and Engineering Mechanics, San Diego State University, San Diego, California 92182

Contributed Papers

4:35

T1. Transient radiation and scattering from fluid-loaded spherical and cylindrical shells. Peter R. Stepanishen (Department of Ocean Engineering, University of Rhode Island, Kingston, RI 02881)

Transient radiation and scattering from fluid-loaded spherical and cylindrical shells has been the subject of numerous investigations during the past 30 years. A new time domain method of addressing these problems is presented here where the velocity field of the shell in fluid is represented as an *in vacuo* modal vector expansion with time-dependent coefficients. For the two-dimensional problems of present interest, sets of two coupled convolution integral equations are developed for the time-dependent modal velocity coefficients. These equations are readily solved by a "marching forward in time" method. The often used plane wave and DAA methods are shown to be special cases of the present formulation. Numerical results are presented for the velocity and pressure field response of a point-excited spherical shell and a plane-wave-excited cylindrical shell. These results will illustrate the transition from light to heavy fluid loading on the transient response of the shells. [Work supported by ONR.]

4:47

T2. Sound radiation from an accelerated vibrating body. Xiao-Feng Wu, Adnan Akay (Department of Mechanical Engineering, Wayne State University, Detroit, MI 48202), and K. Uno Ingard (Department of Physics and Department of Aeronautics and Astronautics, Massachusetts Institute of Technology, Cambridge, MA 02139)

The problem of sound radiation from a moving body is reexamined with particular emphasis on vibrating objects with time-dependent velocity profiles. Specific examples of a rectilinearly accelerating sphere and a vibrating sphere in circular motion are considered. Numerical examples of radiation patterns are given. [Work supported by NSF and WSU Institute for Manufacturing Research.]

4:59

T3. Modeling of the radiation and the transmission of a double stiffened finite plate. J. Nicolas (Groupe d'Acoustique de l'Université de

Sherbrooke, Département de génie mécanique, Université de Sherbrooke, Sherbrooke, Québec J1K 2R1, Canada) and B. Laulagnet (Laboratoire de vibration acoustique, Bâtiment 303, INSA Lyon, 69621 Villeurbanne Cedex, France)

The vibroacoustic behavior of a double finite plate with or without stiffeners and with or without absorbing material is modeled using the modal approach. The excitation of the first plate may be a point force or an acoustic plane wave. The analytical method relies on the following steps. (A) Find the Hamilton's functional of each finite plate including stiffeners effects. (B) Minimize the functional through the projection in the simply supported basis, such as

$$W(x,y) = \sum_{m=1}^{\omega} \sum_{n=1}^{\omega} A_{mn} \sin\left(\frac{m\pi x}{l_x}\right) \sin\left(\frac{n\pi y}{l_y}\right).$$

(C) write the equation of continuity at the vibroacoustic junctions. (D) Describe the coupling between the two plates using propagating waves in the sandwich medium which may be a double layer. (E) Finally, establish the matricial system that permits one to find the displacement amplitude of the second plate. The formulation allows the extraction of some key parameters such as quadratic velocity of each plate, radiated sound power, and transmission loss. Preliminary results for the classical situation will be presented and compared with the case of infinite double plates.

5:11

T4. Coupled finite element/boundary element approach for acoustic radiation and scattering. Gordon C. Everstine, Francis M. Henderson (Applied Mathematics Division 184, David Taylor Research Center, Bethesda, MD 20084), and Luise S. Schuetz (Physical Acoustics Branch 5130, Naval Research Laboratory, Washington, DC 20375)

A new computational capability is described for calculating the sound-pressure field radiated or scattered by a harmonically excited, submerged, arbitrary, 3-D elastic structure. This approach, called NASHUA, first calculates the fluid pressures and velocities on the structural surface by coupling a NASTRAN finite element model of the structure with a boundary element model (based on a discretized form of the Helmholtz surface integral equation) of the surrounding fluid. Farfield radiated pressures are then calculated from the surface solution using the Helmholtz exterior integral equation. The overall capability is very general, highly automated, and requires no independent specification of the fluid mesh. A new out-of-core block equation solver was written so that very large problems could be solved. The use of NASTRAN as the structural analyzer permits a variety of graphical displays of results, including computer animation of the dynamic response. The overall approach is illustrated and validated using known analytic solutions for submerged spherical shells subjected to both incident pressure as well as uniform and nonuniform applied mechanical loads.

5:23

T5. Acoustic radiation from fluid-loaded structures with discontinuities. Charles Thompson and Rahul Sen (Electrical Engineering, University of Lowell, Lowell, MA 01854)

The presence of discontinuities, such as joints, in a fluid-loaded structure results in farfield acoustic radiation even when subsonic wavenumbers prevail in the main body of the structure. The object of this study is to develop a rational model of this phenomenon using singular perturbation methods. When a flexural wave in a thin structure impinges on a joint, evanescent structural displacement fields are set up in the vicinity of the joint. It will be shown that these fields are associated with rotation-dominated modes that are supported by shear-corrected plate theories of the Timoshenko–Mindlin type flexural mode. The method of matched asymptotic expansions will be used to obtain mode-conversion coefficients, and it will be shown that the spacial wavenumber spectrum of the local field is instrumental in setting up an acoustic farfield in the surrounding fluid. An important mathematical issue is the proper description of evanescent fields; the analysis will be based on results obtained previously in connection with acoustic waveguides.

5:35

T6. Acoustic resonances of thin cylindrical shells: Plate waves and the resonance scattering theory. Maryline Talmant, G. Quentin (GPS, Université Paris 7, France), J. L. Rousselot (LCTAR, Vélizy-Villacoublay, France), J. V. Subrahmanyam, and H. Überall (Department of Physics, Catholic University of America, Washington, DC 20064)

An experimental and theoretical study of acoustic backscattering from thin cylindrical shells immersed in water is described. Previous studies showed no general agreement for acoustic scattering from thin elastic shells. In the low-frequency domain ($ka < 150$) and for shells such that the inner–outer radius b/a tends towards unity, two surface waves have been observed: a fast wave that exhibits a dispersion curve similar to that of the S_0 plate mode, and a slow wave. This latter is observed in a limited frequency domain (outside the domain of the S_0 wave), whose extent depends on the value of the ratio b/a for a given material. The previous studies interpreted this wave as the A_0 plate mode, or as Stoneley-type mode. Presented here are relevant experimental results for duraluminum shells with $b/a = 0.96$. A comparison with theoretical calculations from the resonance scattering theory shows good agreement, as well as with the simple model of a solid plane plate. A theory of the physical properties of the slow wave, analyzed as a water-borne wave on a solid plate bounded by water on one side, and by air on the other has been developed.

5:47

T7. The acoustic radiation from a semisubmerged elastic cylinder. François Jouaillec and Guillaume Jacquart (Department of Physics, ONERA, BP 72, 92322 Châtillon Cedex, France)

A large number of techniques have been developed to predict the acoustic radiation of totally immersed elastic structures, but they have not been applied to surface piercing bodies. The case of an infinite elastic cylinder, governed by Donnell's equations, is first considered, its axis lying on the pressure release surface. An analytic theory is derived, and applied to give the radiated field due to a point force excitation. The position of the excitation point with respect to the free surface is discussed as well as the differences between the partially loaded and the totally immersed cylinder. Then a numerical method based on finite elements and integral equations is presented and applied to the case of the partially immersed finite cylinder.

Session U. Speech Communication IV: Neural Networks and Other Techniques (Poster Session)

Ronald A. Cole, Cochairman
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Contributed Papers

All posters will be displayed from 7:30 to 9:30 p.m. To allow contributors the opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 7:30 to 8:30 p.m. and contributors of even-numbered papers will be at their posters from 8:30 to 9:30 p.m.

U1. Noise reduction in speech signal processing by neural network and vector quantization. Kazuo Nakata and Akihiko Sugiura (Tokyo University of Agriculture and Technology, 2-24-16 Naka-machi, Koganei, 184 Japan)

The autocorrelation function provides basic data for speech signal processing by the linear prediction algorithm. The first step toward developing a practical speech signal processing technique is noise reduction in the autocorrelation function of speech corrupted by noise. Recent developments in neural network (NN) techniques have achieved a respectable performance of classifiers even in the worse condition where SNR is less than 0 dB. The vector quantization (VQ) technique, on the other hand, also provides reasonable bases for the discrete description of continuous speech signals. The combination of the two techniques gives the possibility of excellent noise reduction in speech signal processing. Some recent experimental results based on the above principle are given in detail.

U2. Comparison of two LP parametric representations in a neural network-based speech recognizer. K. K. Paliwal (Tata Institute of Fundamental Research, Homi Bhabha Road, Bombay-400005, India)

Although the different linear prediction (LP) parametric representations provide equivalent information about the short-time spectral envelope of speech, these representations are known to show differences in their speech recognition performance when used with conventional linear pattern classifiers. Recently, an error backpropagation algorithm has been reported in the literature for training the artificial neural networks, and it has been shown that the multilayer perceptron (MLP) classifiers that are nonlinear in nature can provide arbitrarily shaped decision surfaces in the multidimensional pattern space. The aim of the present paper is to see whether the different LP parametric representations show differences in their speech recognition performance for these nonlinear MLP classifiers, also. For this, the two-, three-, and four-layer perceptron classifiers are studied for the following two LP parametric representations: (1) the LP coefficient representation and (2) the cepstral coefficient representation. The results for the conventional linear pattern classifiers are also provided for the sake of completeness. It is shown that, like the conventional pattern classifiers, the MLP classifiers also result in better recognition performance for the cepstral coefficient representation than for the LP coefficient representation.

U3. Burst point location in stop consonants using backpropagation neural networks. Shigeyoshi Kitazawa, Mourad Fourati, and Susumu Ichikawa (Shizuoka University, 3-5-1 Jouhoku, Hamamatsu, 432 Japan)

A three-layered neural network approach for burst point location is presented, which can be used to extract consonant segments in a speaker-independent continuous speech recognition system. By using neural networks trained with the backpropagation algorithm [Ruemlhart *et al.*, Nature **323**, 533–536 (1986)], nonlinearity is introduced into the articulatory event detection decision making. The system can detect the burst point location in French voiced stop consonants /b,d,g/. For the experiments, a neural network structure of 12–20 units in the hidden layer, 50 units in the input layer, and 1 unit in the output layer is used. The input patterns represent the time series values of the speech power transition. The network was trained in several steps initially using a smaller set of training data and then using larger sets. The results of the burst location detection were encouraging, especially for the syllables "ba," "dou," and "ga." Generally, the detection rate for /b/ was two times better than that for /d/ and /g/. There was no remarkable difference between the detection rates in the training data and in the unknown data.

U4. Improving the performance of backpropagation-trained vowel classifiers. Gregory R. De Haan (Speech Technology Laboratory, 3888 State Street, Santa Barbara, CA 93105), Omer Egecioglu (Department of Computer Science, University of California, Santa Barbara, CA 93106), and Hisashi J. Wakita (Speech Technology Laboratory, 3888 State Street, Santa Barbara, CA 93105)

Classification experiments using nine steady-state vowels were performed to compare artificial neural networks trained via backpropagation and K-nearest neighbor (KNN) classifiers. Normalized critical-band filterbank outputs served as input patterns in all experiments. Initial experiments used prototypical feedforward networks [R. Lippmann, IEEE ASSP Mag. **4**(2), 4–22 (1987)], with fully interconnected adjacent layers of units. Once the critical number of hidden units [D. J. Burr, J. Acoust. Soc. Am. Suppl. 1 **83**, S46 (1988)] was established for a given experiment, the networks compared favorably to KNN. Significantly, while networks with two hidden layers did better than networks with one hidden layer for (binary) front–back vowel distinctions, they performed worse for (nine-class) vowel classification. It appears that backpropagation may be particularly powerful for binary classification [e.g., R. P. Gorman and T. J. Sejnowski, Neural Networks **1**, 75–89 (1988)]. Experiments were run comparing prototypical networks with partitioned networks, where each hidden unit was connected to only one of the output units. Preliminary results indicate that these partitioned networks, which in effect do nine independent binary classifications in parallel, outperform the prototypical networks, suggesting that the critical number limit on performance can be overcome in a straightforward manner.

U5. Classification of pitch periods using expert knowledge and neural net classifiers. Ronald A. Cole (Department of Computer Science and Engineering, Oregon Graduate Center, 19600 N. W. Von Neumann Drive, Beaverton, OR 97006), Etienne Barnard, Mathew Vea (Department of Electrical and Computer Engineering, Carnegie Mellon University, Pittsburgh, PA 15213), and Fil Alleva (Department of Computer Science, Carnegie Mellon University, Pittsburgh, PA 15213)

A pitch tracking algorithm that uses an artificial neural net (ANN) classifier to identify pitch periods has been developed. The algorithm consists of a peak detector that locates candidate peaks in the filtered waveform (0–700 Hz), and an ANN classifier that uses feature measurements to decide if the candidate peak begins a pitch period. The feature measurements include (a) the amplitude of the candidate peak, (b) the amplitude of the 7 peaks before and the 7 peaks after the candidate peak, (c) the location of these 14 peaks relative to the candidate peak, and (d) the median pitch observed so far in the utterance, based on the prior output of the classifier. The classifier was trained and evaluated on hand-labeled pitch periods in utterances in the DARPA TIMIT database. The classifier contained one hidden layer with six units, and was trained using backpropagation. Initial results on 10 test utterances using small amounts of training data revealed 98% correct classification of candidate peaks. Results of subsequent experiments will be presented.

U6. Location and classification of plosive consonants using expert knowledge and neural net classifiers. Ronald A. Cole (Department of Computer Science and Engineering, Oregon Graduate Center, 19600 N. W. Von Neumann Drive, Beaverton, OR 97006), Etienne Barnard, and Lily Hou (Department of Electrical and Computer Engineering, Carnegie Mellon University, Pittsburgh, PA 15213)

A rule-based segmentation and broad classification algorithm [R. A. Cole and L. Hou, Proc. ICASSP **88**, 453–456 (1988)] located over 95% of segments labeled [b], [d], [g], [p], [t], [k], [ch], [jh], [dh], and [q] (glottal stop) before sonorants in utterances of the DARPA TIMIT database. Artificial neural net (ANN) classifiers were trained to discriminate among the labels using perceptually motivated features. In one condition, 37 feature measurements were used to describe (a) the averaged spectrum during the 15 ms following the release burst, (b) zero crossings and peak-to-peak amplitude contours in the region of the segment, (c) the duration of the segment, and (d) the amplitude of the plosive burst. In a second condition, an additional 16 features were used to characterize the averaged spectrum during the first 30 ms of the sonorant following the plosive. The ANN classifiers consisted of either 37 or 53 input units, 30 hidden units, and 1 output unit for each category. Classifiers were trained using backpropagation and tested on 2000 segments provided by 20 speakers. With different amounts of training, classification accuracy was consistently 3%–5% better when vowel spectra were used, suggesting that ANN classifiers are able to learn coarticulatory relationships between consonant and vowel spectra. Classification accuracy was 70% for the 10 plosive consonants.

U7. Relaxation-based speech labeling. Shigeru Katagiri (ATR Auditory and Visual Perception Research Laboratories, Twin 21 MID Tower, 2-1-61 Shiromi, Higashi-ku, Osaka, 540 Japan)

It was revealed that a trained human spectrogram reader could perform accurate speech labeling, and that accuracy was based on the flexibility of his/her decision process using many kinds of spectrographic features [S. Katagiri, SP87-115, IEICE Tech. Rep. (1988)]. In this paper, a new flexible speech labeling system that simulates the trained reader capability is proposed. The main task of the system is to apply the trial-and-error process used in a human reader's labeling work. Therefore, a relaxation method is adopted here [S. Katagiri, 2-1-19, A. S. J. Spring Meeting (1988)]. The system consists of three parts: an acoustic analyzer, a verifier, and a supervisor. In the acoustic analyzer, many kinds of acoustic feature candidates, e.g., formant and pitch frequencies, are calculated. In the verifier, possible speech labels are verified. The supervisor, with a behavior principle based on the relaxation method, controls the whole system. Experimental results show that the system's performance is comparable to a human reader's performance and that very accurate labels are automatically created.

U8. Incremental learning of large phonetic neural networks from smaller subnets. A. Waibel (ATR Interpreting Telephony Research Laboratories, Twin 21 MID Tower, 2-1-61 Shiromi, Higashi-ku, Osaka, 540 Japan)

Time delay neural networks (TDNNs) [Waibel *et al.*, ATR-TR-0006 (1987)] have recently been shown to achieve excellent recognition performance for difficult phonetic discrimination tasks (e.g., voiced stops). This was achieved in part by the TDNNs' ability to not only activate the correct output category, but also to inhibit all incorrect outputs. The disadvantage of this property is that training larger networks with many more output categories (e.g., all consonants) becomes nontrivial, as all categories have to be incorporated in the learning process requiring excessive amounts of training. Several techniques are presented that overcome this problem by exploiting the hidden structure of previously learned smaller neural nets to train larger nets incrementally in comparatively short training runs. Experimental results show that the resulting larger networks aimed at stop consonants, at voiced stops and nasals, and at all consonants achieve recognition scores (95%–99%) as high as the smaller subnetworks from which they were constructed.

U9. Phoneme recognition using Kohonen's LVQ. Erik McDermott and Shigeru Katagiri (ATR Auditory and Visual Perception Research Laboratories, Twin 21 MID Tower, 2-1-61 Shiromi, Higashi-ku, Osaka, 540 Japan)

Teuvo Kohonen has recently developed an algorithm similar to that used in his feature map classifiers but in which learning is supervised rather than unsupervised. This algorithm, known as learning vector quantization (LVQ), is similar to a K-nearest neighbor algorithm and allows a system to learn the vector quantization of the inputs to different categories. This algorithm is very simple, does not require a large number of training trials, and is capable of forming complex decision regions. As a recognition task, the speaker-dependent recognition of the phonemes /b/, /d/, and /g/ in different phonetic contexts is considered. The training procedure is applied to speech patterns that are stepped through in time, thus providing the system with a measure of shift invariance. Preliminary results indicate that LVQ can yield a recognition rate of 98.3% for 1880 testing tokens from three speakers. The simple vector operations that constitute the core of LVQ allowed for very easy parallelization and thus high learning speed, i.e., less than an hour.

U10. A study of English word category prediction based on neural networks. Masami Nakamura and Kiyohiro Shikano (ATR Interpreting Telephony Research Laboratories, Twin 21 MID Tower, 2-1-61 Shiromi, Higashi-ku, Osaka, 540 Japan)

This paper describes word category prediction based on neural network models for constructing an accurate word recognition system. It

is difficult to represent hidden linguistic structure and make an N-gram word prediction model using traditional stochastic approaches. In this paper, two neural network models that can learn hidden linguistic structure are proposed. These models can easily be expanded from Bigram to N-gram networks. They were tested by training experiments with an open English text database. The Trigram word category prediction rates show that neural network models are comparable to stochastic models. Trigram neural network models compress information about 150 times, which is the ratio of the Trigram stochastic model free parameters ($89^3 = 704\,969$) to the neural network model link weights (4649). In addition, this paper proposes a new method that dynamically controls the training parameters, updating step size and momentum. These techniques are effective for calculating the efficiency of this system.

U11. Connectionist techniques for speaker-independent recognition of isolated utterances. Michael A. Franzini (Department of Computer Science, Carnegie Mellon University, Pittsburgh, PA 15213)

The performance of previous speech-recognition systems has been limited to a large extent by the incomplete structural theories of speech upon which the systems were based. The work reported here suggests that connectionist learning procedures provide an effective method for generating *speaker-independent* recognition systems without the need for any *a priori* model of human speech production. The backpropagation learning algorithm is applied to the problem of isolated-utterance speech recognition, and the resulting recognition rates approach those of the best conventional systems. Several studies were peformed using computer simulations of various networks trained by backpropagation. With 1 s of digitized speech as input, the networks had to generate as output the appropriate labels, which in these studies were letters of the alphabet. The accuracy for speaker-independent recognition of the whole alphabet reached 89%, which is higher than the accuracies achieved by several more conventional recognizers using the same database [K. F. Lee, Carnegie Mellon Univ. Tech Rep. 85-181 (1985)]. Recognition rates for speaker-dependent recognition of the whole alphabet reached 99% and, for speaker-independent recognition of confusable letter sets such as b–p–e–v–d, the networks achieved 94%. These studies demonstrate that networks with simple task-independent learning procedures can perform as well as systems that explicitly implement procedures such as dynamic time warping and vector quantization of inputs.

U12. Speaker identity feature combination with a neural net. George Velius (Bellcore, 445 South Street, Room 2E-244, Box 1910, Morristown, NJ 07960-1910)

This study compares the performance of a neural net approach with conventional linear methods as applied to the problem of feature combination in the domain of speaker identity verification (SIV). The experiment endeavors to combine features consisting of LPC-cepstral coefficient differences and pitch differences for isolated words in a template-matching scenario. The signal features are analyzed for 30-ms frames every 10 ms. The pitch estimate is based on the cepstrum of the LPC residual. Previous work [G. Velius, ICASSP **88**, 583–586 (1988)] showed that the Fisher linear discriminant (FLD) was better at feature weighting (for cepstral coefficients only) than several other common linear methods. Results show that, when feature combination is done by the neural net, the SIV task is performed significantly better than when the feature combination (i.e., weighting) is done by the FLD. The neural network architecture used in this experiment was in no way "optimized" for the specific task at hand. An additional finding is that the pitch feature used here, in conjuction with the cepstral coefficients, contributes significantly to the SIV task; that is, the error rate is reduced by 13%.

U13. The LPC trace as an HMM development tool. George R. Doddington, Joseph Picone, and John J. Godfrey (Speech Research Branch, Computer Science Center, Texas Instruments Incorporated, P. O. Box 655474, Mail Stop 238, Dallas, TX 75265)

Hidden Markov models have gained wide acceptance in speech recognition due to the ability to construct optimum (maximum likelihood) models automatically from speech data. Understanding how recognition performance is related to a model's structure is not a simple matter, however, especially where the structure is complex. Thus analysis of errors, especially in terms of the properties of the speech data, is not often undertaken. This paper describes a method for relating speech recognition performance to HMM structure. The basic tool is a best-path (Viterbi) trace of the model through the input data, which is represented using well-known LPC parameters. Linear predictor parameters are used as auxiliary model parameters, independent of the HMM output parameters used for recognition, in order to take advantage of established LPC speech synthesis and LPC speech spectrogram utilities. The trace can then be used to gain insight into error mechanisms, often leading to improvements in model structure and system performance. The description will include techniques for creating the LPC auxiliary model, spectrographic and auditory output from supervised and unsupervised model traces, and an evaluation of the impact of this technique on the development of an HMM-based speech recognition system.

U14. Improved training procedures for hidden Markov models. Lawrence R. Rabiner, Chin-Hui Lee, Biing-Hwang Juang, David B. Roe, and Jay G. Wilpon (Speech Research Department, AT&T Bell Laboratories, Murray Hill, NJ 07974)

Techniques for training hidden Markov model (HMM) parameters from a labeled training set of data are well established and include the forward–backward algorithm as well as the segmental K-means algorithm. These algorithms have been shown to be capable of estimating the parameters of an HMM based on mathematically well-founded techniques. In practice, however, difficulties are often encountered when estimating some of the HMM parameters. These difficulties are generally the result of having insufficient training data to give robust and reliable parameter estimates. Typically, the model parameters most affected by having insufficient training data are the spectral parameter variance estimates, and the estimates of parameters related to the modeling of state duration. Although techniques have been proposed for improving estimates of the variances due to the effects of insufficient training data, the results have not proven adequate in some cases. As such, improved training techniques (which give better recognition performance) have been devised for controlling the minimum variance estimate of any spectral parameter, and for thresholding and clipping state duration parameter estimates. These improved training methods have been tested on several databases with good success. In addition, advanced techniques for creating multiple HMMs from the training data (i.e., for speaker independent recognition) have been devised and have proven successful for modeling large databases of training material.

U15. Duration control methods for HMM phoneme recognition. Toshiyuki Hanazawa, Takeshi Kawabata, and Kiyohiro Shikano (ATR Interpreting Telephony Research Laboratories, Twin 21 Building MID Tower, 2-1-61 Shiromi, Higashi-ku, Osaka, 540 Japan)

Two kinds of duration control for HMM (hidden Markov model) phoneme recognition are proposed: phoneme duration control for an HMM phone model and an event duration control for an HMM state. The phoneme duration control is carried out by combining an HMM output probability with a phoneme duration probability. The phoneme duration probability is calculated using a phoneme duration histogram obtained from training samples. Phoneme duration control is effective in discriminating phonemes with different durations such as /n/ and /N/. Event duration control is realized as a state duration penalty calculated from an HMM state duration probability of training samples. Event duration control is effective in discriminating phonemes with different event structures such as /s/ and /ts/. Recognition experiments are carried out using Japanese phonemes extracted from an isolated word database uttered by one male speaker. The phoneme recognition rate is improved from 84.8%–90.0% using these duration control techniques.

U16. Speaker adaptation in HMM. Yasuhisa Niimi and Yutaka Kobayashi (Department of Computer Science, Kyoto Institute of Technology, Matsugasaki, Sakyo-ku, Kyoto, 606 Japan)

A method for speaker adaptation of a code book in vector quantization and its application to word recognition based on the HMM are reported. Under the assumption that speech vectors can be represented by the two-factor model—that is, a sum of the two main effects of "phoneme" and "speaker" and the interaction between the two—the vector space is divided into narrow subspaces in which the speaker effect is considered constant. In each subspace, the displacement vector due to that effect is estimated by using training utterances. The speaker adaptation of a reference code book is completed by moving all the code vectors contained in the subspace parallel to the displacement. The recognition of 65 Japanese city names was performed. A speaker-independent reference code book and the HMMs of the words were designed by using utterances spoken by five male speakers. For the utterances produced by 20 other male speakers, the average word recognition rate was 97.4% in the speaker-adaptive mode and 94.1% in the speaker-independent mode.

U17. Hidden Markov models for Spanish stops. Horacio Franco (Laboratorio de Investigaciones Sensoriales, CC 53 (1453), Buenos Aires, Argentina)

In this work, the design and evaluation of the recognition performance of hidden Markov models (HMM) for the intervocalic Spanish stops are described. The proposed HMMs were context dependent in order to account for coarticulatory effects. Continuous probability density functions were used for the output vectors, which included spectral dynamic parameters. Initial model parameter estimates were obtained by means of an *ad hoc* segmentation procedure for careful modeling of spectral transitions. The speech database consisted of a total of 2592 productions of the Spanish stops /p, t, k, b, d, g/ in intervocalic position with the vowels /a, i, u/ in all combinations. The speech data were produced by two male Argentine speakers. As a step to evaluate the recognition performance with running speech, three different consonants were produced embedded in VCVCVCV nonsense utterances. Results were obtained from variations of the basic models that consisted of a comparative study of the recognition performance under different degrees of context dependence, and the alternative use of spectral transition parameters. In the best condition the average consonant recognition performance, obtained in a speaker-dependent manner, was 96.61% and 96.53% for each speaker. The errors were concentrated mainly in the voiced stops, with almost perfect recognition obtained for the unvoiced stops.

U18. Speaker-independent phoneme recognition using hidden Markov models. Kai-Fu Lee and Hsiao-Wuen Hon (Department of Computer Science, Carnegie Mellon University, Pittsburgh, PA 15213)

In this paper, the currently popular hidden Markov modeling to speaker-*independent* phoneme recognition is extended. Using multiple code books of various LPC-derived parameters and discrete HMMs, speaker-independent phoneme recognition accuracy of 58.8%–73.8% on the DARPA TIMIT database, depending on the type of acoustic and language models used, is obtained. In comparison, the performance of expert spectrogram readers is only 69% without use of higher level knowledge. The *co-occurrence* smoothing algorithm that enables accurate recognition with only a few training examples of each phone is also introduced. Since these results were evaluated on a standard database, they can be used as benchmarks to evaluate future systems. [Work supported by DARPA.]

U19. Word spotting method based on HMM phoneme recognition. Takeshi Kawabata, Toshiyuki Hanazawa, and Kiyohiro Shikano (ATR Interpreting Telephony Research Laboratories, Twin 21 MID Tower, 2-1-61 Shiromi, Higashi-ku, Osaka, 540 Japan)

A new technique for detecting and locating key words in continuous speech using (hidden Markov model) (HMM) phoneme recognition is proposed. HMM word models are composed of HMM phone models trained on an isolated word database. Because the speaking rates for isolated words and continuous speech are different, phoneme spectra and durations change considerably. An HMM consists of several states and arcs. Each arc has output probabilities for each VQ code. In order to cope with the spectral changes, the output probabilities are smoothed with the probabilities of their spectral neighbor codes. In order to cope with the duration changes, HMM state duration parameters are shifted according to a second-order duration calibration curve. The calibration curve is obtained from a speaking rate ratio of continuous speech to isolated words. The word detection rate for 8 key words in 25 sentences uttered by one speaker was 98.4%. Accurate word spotting is accomplished using the HMM output probability smoothing technique and the state duration control mechanism taking the speaking rate into account.

U20. Estimating the acoustic characteristics of speech from visual speech signals. Ben P. Yuhas (Speech Processing Laboratory, Department ECE, Johns Hopkins University, Baltimore, MD 21218)

Speech articulation produces acoustic and visual signals. When the acoustic signal is degraded by noise, the visual signal can provide compensatory information [W. H. Sumby and I. Pollack, J. Acoust. Soc. Am. **26**, 212–215 (1954)]. The aim of the work to be presented is to explore the extent to which the transfer function of the vocal tract filter can be estimated from the visual speech signal. The approach was to use a neural network to obtain the short-term power spectral envelope of the acoustic signal given the corresponding visual image as input. The visual images were taken from laser disc recordings of speaker's faces [Bernstein *et al.*, J. Acoust. Soc. Am. Suppl. 1 **82**, S22 (1987)]. A small box centered at the mouth was extracted and spatially subsampled. The output of the neural net was the acoustic spectral envelope of the corresponding acoustic signal obtained via the cepstral method. After being trained on 95 tokens of vowels and dipthongs, the network was tested on 24 images it had not seen. The net was able to estimate the spectral envelope associated with these images more accurately than more traditional approaches that used stored templates. [Work supported by AFOSR Contract No. 86-0246.]

U21. A TRACE-like speech perception simulation model for the recognition of Japanese phonemes. Meinrad Niemöller, Shozo Makino, and Ken'iti Kido (Research Center for Applied Information Sciences, Tohoku University, 2-1-1 Katahira, Sendai, 980 Japan)

Based on the concept of the interactive activation model of speech perception (TRACE model) introduced by McClelland and Elman, a simulation model for the recognition of Japanese vowels and voiced plosives has been built up. The system currently consists of a feature and a phoneme level. For preprocessing, the output of a 29-channel filter bank is reduced by KL transformation to 16 dimensions and presented to the feature level every 10 ms. The phonemes are represented by nodes every 20 ms and have excitatory connections to five and to nine feature slots for consonants and vowels, respectively. In each processing cycle, there is lateral inhibition of nodes of the same level, feedforward and feedback excitation between the levels, and an activation update. A stable phoneme response output is reached after about 30–40 processing cycles. For the training of the feature-phoneme mapping by a gradient-descent procedure, a database of 212 words uttered by ten male speakers is used.

U22. NNet: A natural voice neural network speech synthesizer. James A. Villarreal, Gary McIntire [Artificial Intelligence Section/FM 72, Lyndon B. Johnson Space Center (JSC), NASA, Houston, TX 77058], Claire W. Anderson, and Wallace L. Anderson (Electrical Engineering Department, Cullen College of Engineering, University of Houston, Houston, TX 77004)

Reproduced speech extracted from actual voice recordings of a few selected individuals representing a wide variance of age, speech dialect (English), and sex was used as input values to a four-layer (input, hidden, output, and state) generalized backward propagation artificial neural network system (ANS) using a linear prediction coding (LPC) model to produce more natural sounding artificial speech. The speech phonetic research environment (SPIRE) MIT-developed software package running on a Symbolics computer was utilized in making the original voice recordings. The linear predictive analysis of the LPC model efficiently represented the speech signals in terms of slowly varying parameters. It converted the combined spectral contributions of the glottal flow within a pitch period, the vocal tract, and the radiation of the lips into a single recursive (all-pole) time-varying filter. The transfer function of the filter involved the use of gain coefficients that, when fed into the system, were found to destabilize it. The LPC model was modified to use impulse characteristics of the system rather than the individual discrete coefficients. Preliminary data indicated close output to input matches, artificial speech to natural speech, indicating the ability of the artificial neural network to learn the timing, pitch fluctuations, and connectivity between individual sounds, and the speaking habits unique to a person.

Session V. Architectural Acoustics IV: Reverberant Sound Fields in Rooms

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Contributed Papers

8:00

V1. Sound power determination in reverberation chambers. Dah-You Maa (Institute of Acoustics, Academia Sinica, Beijing, P. O. Box 2712, People's Republic of China)

The determination of sound power in reverberation chambers is discussed in the light of normal mode theory. It is known that the sound power emission of a source depends on its position in the reverberation chamber and on interference pattern results in the sound field excited. As a consequence, the measured sound pressure varies widely from point to point. It is shown that the sound pressure is proportional to the average contribution of the product of the normal functions at the source and at the receiver. Based on this relation, the determination of sound power emitted by the source at a particular point in the room is devised through the measurements of average sound field as well as by a corner microphone. It is also shown that sound power determined in reverberation chambers by the standard method is always less than the free-field power, and the difference increases as the frequency decreases. This is just what happens in practice, and good agreement with the theory is obtained with earlier experiments. A statistical formula of the sound pressure in a reverberation chamber developed from the exact theory is used to this end.

8:12

V2. Steady-state acoustic energy distribution in a reverberation chamber at low frequencies. Daniel R. Flynn, Thomas W. Bartel, and Simone L. Yaniv (National Bureau of Standards, Gaithersburg, MD 20899)

The spatial variation of the mean-square sound pressure in a hard-walled rectilinear reverberation chamber is analyzed by extending the "Waterhouse theory" (based on a free-wave model for an omnidirectional sound field impinging on the corner formed by three orthogonal planes) to apply to a closed chamber at low frequencies. This is done by expressing the mean-square sound pressure in terms of the contributions from the actual pressure microphone and from an infinite array of image microphones due to multiple reflections in the chamber walls, and then transforming this expression into a weighted sum of the normal eigenfunctions for the chamber. Experimental sound-pressure levels, measured along different linear paths in the 425-m^3 NBS reverberation chamber, are compared with the predictions of this analysis. It is found that the spatial dependence of sound-pressure level is accurately predictable, at least in the NBS chamber, to significantly lower frequencies than has usually been thought possible. Expressions are given for relating the spatial average (over the volume of the chamber) mean-square sound pressure to that which is measured at a given position in the room. The implications of these findings to sound power determinations are discussed.

8:24

V3. Kinetic and potential energies of a stationary sound field. Richard K. Cook (4111 Bel Pre Road, Rockville, MD 20853)

There is a continuing need for accurate measurement of radiated sound power by compact noisy sources. Examples are the noises of fans, motors, machinery, etc. One of the standard methods of measurement makes use of a reverberation chamber. The potential energy of the sound radiated into the chamber is measured by sampling the sound pressure at various points, from which the time-space-averaged potential energy is calculated. The kinetic energy is assumed to equal the potential energy; then in the standard method, the sum of the two is used as the total energy for calculation of the radiated sound power. In experiments on a source having an accurately known radiated power, we found that the power determined in a standard-method chamber was about 20% (0.9 dB) less than the true power. A likely source of the systematic difference is the possibly false assumption that the time-space averaged kinetic energy is equal to the potential energy. It is shown that, in general, in a sound field stationary in an enclosure, the two energies are not equal. Several examples are presented. Included are fields in enclosures having sound absorbers. An accurate ratio of the energies requires detailed knowledge of the field.

8:36

V4. Impulse response and reverberation time measurements using a periodic pseudorandom sequence. W. T. Chu (Acoustics Section, Institute for Research in Construction, National Research Council of Canada, Ottawa, Ontario K1A 0R6, Canada)

The cross-correlation method using a periodic pseudorandom sequence or *M* sequence for impulse response and reverberation time measurements proposed by Schroeder and his colleagues [H. Alrutz and M. R. Schroeder, Proc. 11th Congr. Acoust., Paris (1983)] represents a very significant contribution to the fields of Architectural Acoustics and Noise Control. With this technique, it is not only possible to make measurements in auditoria or offices without disturbing the audience or the occupants, but also in factories or construction sites without closing down their operations. Implementation of this method requires special signal processing techniques which will be reviewed in this presentation. Examples of measurements in simulated noisy environments will also be presented.

8:48

V5. Analysis of sound fields in rooms by Bergeron's method. Hidemaro Shimoda (Institute of Technology, Shimizu Corporation, 3-4-17 Etchujima, Koto-ku, Tokyo, 135 Japan), Norinobu Yoshida, and Ichiro Fukai (Faculty of Engineering, Hokkaido University, N13-W8 Kita-ku, Sapporo, 060 Japan)

The behavior of sound wave motion in rooms is so complicated that it is not easy to treat it theoretically unless simple geometrical shapes or simple boundary conditions are assumed. This study presents a formulation applying Bergeron's method, developed for an electromagnetic field simulation by computer [N. Yoshida *et al.*, Trans. IECE Japan, J62-B, 6, 511–518 (1979)], to room acoustics. In this formulation, the sound field is represented by an electrical equivalent circuit composed of distributed lines corresponding to the acoustic equation, and nodal equations for each connection of the lines are derived. As an example, consider a cube-shaped room with uniform absorption by each wall. The transient responses to the sinusoidal waves and the 1/3-octave-band tone burst are calculated, and the stationary sound-pressure distribution and the reverberation time are obtained from these responses. The results show the validity of the formulation and prove the effectiveness of the application of this method to room acoustics.

9:00–9:05
Break

9:05

V6. Measurements of short reverberation times using a real time analyzer. Klaus Højbjerg (Brüel & Kjaer, Naerum, Denmark)

Classical measurements of reverberation time with use of the interrupted noise method is known to almost everybody working with acoustics. However, there seems to be less knowledge about the limitations of the method, i.e., the connection between measurement bandwidth and minimum averaging time and what that means for minimum reverberation time measured. Another method is impulse excitation followed by a backwards integration. Here, the analysis can be done in two ways, either direct frequency analysis of the response signal giving the same limitations as with the interrupted noise method or by recording the time signal and then feeding the signal reversed to the filter bank giving possibilities of measuring very short reverberation times. The methods of measuring reverberation time will be discussed and examples of measurements shown.

9:17

V7. Frequency distribution of normal modes of vibration in rectangular rooms: A new look. Ludwig W. Sepmeyer (Consulting Engineer, 1862 Comstock Avenue, Los Angeles, CA 90025)

In a previous study on the frequency and angular distribution of the normal modes of vibration in rectangular rooms, published in 1965, the criterion for judging the frequency distribution of the modes was called the mode spacing irregularity MSI. The MSI, based on the ratio of the actual mode spacing between two adjacent modes to the ideal spacing between two modes predicted by Maa's equation, is quite insensitive to changes in room dimension ratios. In this paper, a new criterion called the modal volume distribution index, MVDI, is proposed. The MVDI is based on the difference in spherical volume enclosed by two adjacent modal vectors in frequency space and the difference in volume between two similar ideal vectors. In this way, larger and smaller than ideal spacings are weighted equally. Results, quite different from those inferred from the MSI, are given for a few selected dimension ratios. Revisions to National and International Standards that require sound-pressure level measurements in reverberation rooms are recommended.

9:29

V8. Calculation of sound radiation from an aperture of a building by means of Kirchhoff's diffraction formula. Nobuo Hara and Yoshihiro Furue (Department of Architectural Engineering, Kyoto University, Sakyo-ku, Kyoto, 606 Japan)

When a noise source is located in a very large room as compared to the wavelength concerned, the integral equation method to calculate the sound field is not applicable because of limitations in computer capacity. In this case, an approximate approach based on Kirchhoff's diffraction formula becomes more practical. This formula requires the velocity potential and its normal derivative at the aperture, which are unknown. These unknowns may be assumed to consist of the direct wave from the source and the diffused ones. The latter are assumed to be plane waves with the same amplitude but random phase. The relation between the amplitude of the direct wave and the reflected ones is determined by geometrical acoustics. Thus the radiated sound field can be readily evaluated from the aperture by Kirchhoff's formula. The numerical results for a scale model of a reverberation chamber with a rectangular aperture are shown with measurements. The agreement between the calculations and the measurements is satisfactory.

WEDNESDAY AM

9:41

V9. Analysis of the sound field in a room with uneven surfaces by the hybrid method combining geometrical theory and wave theory. Kyoji Fujiwara (Department of Acoustic Design, Kyushu Institute of Design, 4-9-1 Shiobaru, Minami-ku, Fukuoka, 815 Japan)

This study presents a new way of estimating the sound field in a room with walls that do not reflect the incident sound specularly. So far the reflection characteristics of these walls have been considered to be perfectly diffuse or a mixture of partially diffuse and partially specular. The aim of this study is to introduce the real reflection characteristics of the walls into the simulation procedure for the sound field. The estimation procedure is essentially based on the Monte Carlo simulation method. For simplicity, the uneven walls are assumed to be periodically corrugated. Prior to the simulation, the reflection characteristics of the periodically corrugated wall are analyzed by wave theoretical methods, and the result, which contains the directions and magnitudes of the reflected waves, is used in the simulation procedure. The energy impulse responses estimated by the method presented are compared with those obtained by the ordinary method based only on geometrical theory.

Session W. Engineering Acoustics III: Modern Acoustic Transducers

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Juro Ohga, Cochairman
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1015 Kamikodanaka, Nakahara-ku
Kawasaki, 211 Japan

Chairman's Introduction—8:00

Invited Papers

8:05

W1. New developments in the field of acoustic transducers. G. M. Sessler (Technische Hochschule Darmstadt, Merckstrasse 25, D-6100 Darmstadt, West Germany)

New developments in the field of acoustic transducers based on the silicon, fiber-optic, and polymer technologies are reviewed. The silicon transducers are produced on silicon wafers with micromechanic methods utilizing anisotropic etching procedures. Subminiature transducers of this kind, based on the capacitive or piezoelectric principles, were made with membranes of about 1-mm^2 area and less than 1-μm thickness. Their use as "intelligent sensors" with on-chip signal-processing capability is presently under study. Fiber-optic sensors consist of glass fibers and detect modulations of either phase or amplitude of the transmitted light waves, caused by interaction of the sound waves with the fiber. Such transducers have initially been used as hydrophones and accelerometers but are now also under study as microphones. Polymer-electret transducers are finding widespread applications as microphones, earphones, and ultrasonic devices. New developments include the use of electret biasing in silicon transducers and the design of electret-microphone arrays and antennas. Piezoelectric polymer transducers were recently improved by the use of better materials and by innovative poling techniques, such as methods to produce single-film monomorphs and bimorphs.

8:35

W2. Fiber-optic sensors for vibro-acoustic measurements. S. L. Garrett (Physics Department, Naval Postgraduate School, Monterey, CA 93943)

Optical fiber sensors of sound and vibration offer potential advantages over conventional (electronic) sensing techniques in applications where there is strong electromagnetic interference, large sensor/receiver separations, or restrictive (remote) power requirements. The rapid acceptance of optical fibers as a communications medium over the past decade has stimulated much research in fiber-optic sensing but has generated very few commercially available vibro-acoustic sensors of the intensity [J. W. Berthold, "Overview of fiber-optic intensity sensors for industry," in *Fiber Optic and Laser Sensors V*, Proc. SPIE **838**, 2–8 (1987)] or the interferometric [C. M. Davis, "Fiber optic sensors: An overview," Opt. Eng. **24**(2), 347–351 (1985)] types. This presentation will review the basic fiber-optic sensor types (amplitude/phase and intrinsic/extrinsic), present examples of microphone, hydrophone, and accelerometer designs, and speculate on the factors that will influence their commercialization over the next decade. Issues of minimum detectable signals, electro-optic demodulation, and multisensor array multiplexing will be addressed as time permits.

8:55

W3. Piezoelectric acoustic transducers for electronic telephone sets. Juro Ohga (Fujitsu Laboratories, 1015 Kamikodanaka, Nakahara-ku, Kawasaki, 211 Japan)

The development of piezoelectric ceramic transducers for the transmitter, receiver, and tone-ringer sounder of recent electronic telephone sets is reviewed. As the first development stage, the working attenuation (WA) was studied as a measure to compare efficiencies of various types of electroacoustic transducers. The conclusions were that WA for electroacoustic transducers, which was defined in a similar way to WA for electrical circuits, could be estimated easily from the specific response for a microphone, and that WA was almost independent from transducer size. Then, it was shown by comparison of WAs that the efficiency of the simplest piezoelectric ceramic transducer using a unimorph diaphragm was as high as that of ordinary electromagnetic or electrodynamic ones. Moreover, a piezoelectric transmitter and receiver needed the same resonant frequency of the diaphragm, because both diaphragms should work in the stiffness-controlled region. Fortunately, their optimum resonant frequency was close to that of the tone-ringer sounder. Therefore, the main problem of the second development stage was how to design acoustical circuits for the piezoelectric transmitter, receiver, and tone-ringer sounder using the same diaphragms. This was solved by introducing a circuit with four degrees of freedom. As a related problem, simplification of transducer structure was also studied.

9:15

W4. A new piezoelectric microphone with divided electrodes. Nobuomi Imai (Department of Research and Development, Primo Co., Ltd., 6-25-1 Mure, Mitaka, 181 Japan)

A new type of piezoelectric ceramic microphone is proposed. Its merits are as follows. (1) High output, high efficiency—A high output voltage was obtained by dividing the piezoelectric ceramic electrodes and connecting them in series. The output voltge is proportional to the number of divisions. However, since the impedance is high, an FET circuit is required to decrease output impedance. At the same time, an improvement of the signal-to-noise ratio was also realized. (2) Reduction of diaphragm mechanical impedance—To improve the matching to the impedance of air, a mechanical transformer was introduced. The center of the diaphragm was lightened and its rigidity was increased by combining a dome-type diaphragm and ring-shaped piezoelectric ceramic. This made it easy to control the frequency response. (3) Stability—The diaphragm consists of a $Y_2O_3 \cdot$PSZ ceramic thin film formed into a dome shape and laminated to a ring-shaped ceramic. This structure showed a higher temperature stability and noticeably lower aging change than the conventional flat-type diaphragm. The specifications of the prototype microphone are the following: diameter, 1 in. (23.8 mm), nondirectional; sensitivity, -59 dB/μbar at FET source output; frequency response, 20 Hz–12.5 kHz $\pm$ 1.5 dB; S/N ratio, 59 dB A-weighted.

9:35–9:45
Break

WEDNESDAY AM

9:45

W5. Use of thin PVDF film for measuring small single-crystal samples of high temperature superconductors. J. D. Maynard, J. H. Mather (The Pennsylvania State University, University Park, PA 16802), Stewart Brown, and Albert Migliori (Los Alamos National Laboratory, Los Alamos, NM 87545)

Although polyvinylidene fluoride (PVDF) piezoelectric film has been adopted in commercial applications for some time, its use in basic research has been less extensive. Recently, there have been found significant advantages in using PVDF to study the ultrasonic properties of the new high-temperature oxide superconductors. In order to probe the anisotropy of the oxide superconductors, it is necessary to study single crystals, but currently available crystals are very small, on the order of the millimeter square and only about 100 μm thick. Measuring the temperature dependence of the ultrasound in such a small sample without having the transducer (or its leads or substrate) intrude on the measurement is a difficult problem that is not readily solved with conventional transducer materials. However, PVDF films as thin as 9 μm are commercially available and easily adapted to small samples and low-temperature operation. Small active areas and leads are produced with metalization patterns on each side of the PVDF film. For resonance measurements, electrical crosstalk across the small sample is processed by frequency modulating the drive and using phase sensitive detection. In our experiment, resonances in $\simeq$ 100-μm samples are measured with large signal-to-noise ratios. [Work supported by the Office of Naval Research and NSF Grant DMR 8701682.]

10:05

W6. Pulse discharge sound source and its application to acoustic measurement. Hideo Shibayama (Department of Electrical Communication, Shibaura Institute of Technology, 3-9-14 Shibaura, Minato-ku, Tokyo, 108 Japan) and Ken'iti Kido (Research Center for Applied Information Sciences, Tohoku University, 2-1-1 Katahira, Sendai, 980 Japan)

This paper describes a pulse discharge sound source without any diaphragm, and its application for acoustic measurements. The waveform of the sound pulses radiated from this source is simple and reproducible. The power spectra of these pulses have no zero points up to 90 kHz. The duration of the sound pulse is about 50 μs, and is dependent on the gap length between the discharge electrodes. The directivity function of the sound source is the same as that of a line source that is equal to the gap length. The sound generating spot is small, and the sound source requires no baffle board. As a result, no reflection from the sound source itself is produced. The timing of sound generation is stable and controllable. As examples of acoustic measurements utilizing the sound source, the estimated results of acoustic impedance characteristics of materials and their frequency responses of diffraction are shown. The estimated results agreed well with the calculated ones. A measuring system can be considerably simplified by use of the pulse discharge sound source and the digital signal processing technique. Measuring time can also be reduced.

10:25

W7. Directional microphone systems for teleconferencing. G. W. Elko and J. E. West (Acoustics Research Department, AT&T Bell Laboratories, 600 Mountain Avenue, Murray Hill, NJ 07974)

Many factors have contributed to historically unrealistic predictions for the growth of audio teleconferencing. Aside from the sociological issues, there exist many acoustic problems that can severely degrade the

performance of present technology. A few of these problems are: the effects of reverberation, room noise, feedback, undesired speakers, and echo. At present, the most common solution for some of these problems has been to locate a microphone(s) in close proximity to the desired speaker(s) with gain switching to eliminate feedback problems. While this solution can sometimes be effective, it also introduces some compromises that are easily spotted and are considered annoying by teleconferencing users. The most notable problems are with gain switching and reverberation. These detrimental effects can be reduced by the use of highly directional microphones systems tailored to the problems of teleconferencing. The development of directional microphones and their present use in audio teleconferencing will be reviewed. Some of the highly directional microphone arrays that have been worked with will be described. Finally, comments will be given on how these new directional microphone systems address the problems of teleconferencing in small and large acoustic environments.

10:45

W8. Modern techniques for frequency response measurement of loudspeakers. Shokichiro Yoshikawa (Kanagawa Institute of Technology, Atsugi, 243-02 Japan) and Takashi Wakuri (Science and Technical Research Laboratories of NHK, Setagaya-ku, Tokyo, 157 Japan)

Usually, the frequency responses of loudspeakers are measured by the point-by-point method in a free sound field. In this method, accuracy of the measurement depends on the precision of the free sound field. Since the wavelength of sound is quite long in the low-frequency range, an anechoic room of a large size with high accuracy is required to obtain good results in the low-frequency range. To overcome this difficulty, several new measuring methods have been proposed [e.g., J. Merhaut, J. Audio Eng. Soc. 30, 882 (1982); and IEC84/WG8 (Chairman) 1 April (1984)]. In this study, a comparison and evaluation of new measuring methods were carried out. Factors that affect the accuracy of measurement were investigated. The accuracy and the applicable frequency range were clarified. The possibility of measuring the frequency response of loudspeakers without using an anechoic room will be discussed.

Contributed Papers

11:05

W9. Application of piezoelectric composite for large-area hydrophone arrays. Fred G. Geil (Westinghouse Electric Corporation, Oceanic Division, P. O. Box 1488, Annapolis, MD 21404) and Robert Y. Ting (Naval Research Laboratory, Underwater Sound Reference Detachment, P. O. Box 568337, Orlando, FL 32856-8337)

Large-area hydrophone arrays using flexible materials are subject to high levels of interelement coupling arising from the material in-plane sensitivity coefficient g_{31}, in the presence of flexural energy within the array module. A "0–3" type piezocomposite offers a solution to this interelement coupling problem without introduction of fabrication complexity such as stiffening of elements. A simplified array fabrication concept will be presented, as well as results from small array tests using commercially available "0–3" piezocomposites based on lead titanate powder dispersed in a rubber matrix. These test results show a near-zero level of acoustic contamination, shown by directivity patterns and relative phase responses. Besides low interelement coupling, cost, waterproofing, and electrical interfacing are also important in the design considerations of affordable, realistic arrays for the seawater environment. The advantages of using "0–3" piezocomposites in meeting these additional requirements will be briefly discussed. [Work supported by ONT.]

11:17

W10. A balanced, centrally stiffened design for PVDF hydrophone arrays. Robert Y. Ting (Naval Research Laboratory, Underwater Sound Reference Detachment, P. O. Box 568337, Orlando, FL 32856-8337) and Fred G. Geil (Westinghouse Electric Corporation, Oceanic Division, P. O. Box 1488, Annapolis, MD 21404)

As PVDF thick films become more readily available, there has been a growing interest in the development of a large-area hydrophone for passive sonar applications. These new PVDF materials offer the advantages of having a lower density, greater mechanical strength and flexibility, and better impedance match to water than conventional piezoelectric ceramics. Furthermore, they are now available in large sheets, from which arrays can be developed for acoustic detection covering large surface areas. Prototype PVDF hydrophone arrays were fabricated and tested to show that in-plane flexural motion greatly affected hydrophone responses due to the in-plane anisotropy of the piezoelectric properties of PVDF. To circumvent this difficulty, a balanced, centrally stiffened configuration was adapted by sandwiching two pieces of PVDF to a common damped plate, while aligning in parallel the stretched "3–1" directions of the PVDF sheets. With this design, constant receiving sensitivity and directivity patterns in agreement with theoretical predictions were obtained. Further refinement of this design was also achieved through mathematical modeling. [Work supported by ONR.]

11:29

W11. Comparison of ceramic and thick-film PVDF hydrophones for large surface arrays. B. Tocquet, B. Fromont, and M. Josserand (DSA–Thomson-Sintra ASM, 525, route des Dolines Parc de Sophia Antipolis, BP 38, 06561 Valbonne Cedex, France)

A special process allows the manufacturing, in industrial conditions, of thick, nonvoided PVDF films (up to 2 mm) with excellent piezoelectric properties. The characteristics of PVDF are compared with those of some classical PZT ceramics. From such films, very simple and reliable large surface sensors have been designed and developed. Compared to classical ceramic hydrophones, these sensors show a great stability versus temperature and pressure and a very good sensitivity to self-noise ratio for low-frequency passive listening systems. Furthermore, these large surface sensors fit very well with hard baffles and are less sensitive to flow noise than small ceramic hydrophones. Some examples of results obtained with such new hydrophones are given. Clusters of such PVDF hydrophones can be encapsulated with viscoelastic "ρc" materials in flat panels and fixed, with decoupling layers or directly on the hull of the ship, in direct contact with the hydrodynamic flow, leading to a very compact and reliable technology for large flank arrays.

11:41

W12. On the use of terfenol D in underwater acoustic projectors. Mark B. Moffett, Jan F. Lindberg, and James M. Powers (Naval Underwater Systems Center, New London, CT 06320)

Terfenol D, an alloy of iron, terbium, and dysprosium, is a highly magnetostrictive material that can provide significantly more power than lead zirconate titanate (PZT) under field-limited conditions. Field-limiting at resonance, however, requires a mechanical quality factor Qm, less than unity for terfenol D and less than four for PZT. Transducers that are a small fraction of one wavelength in size will have to be elements of a large array in order to achieve the requisite low Qm's. If the Qm is too high, the transducer will be stress-limited, rather than field-limited, and the use of terfenol D is not particularly advantageous. [Work supported by NUSC's IR/IED program and by ONT.]

WEDNESDAY MORNING, 16 NOVEMBER 1988

OAHU/WAIALUA ROOM, 8:00 A.M. TO 12:04 P.M.

Session X. Physical Acoustics III: Acoustic Microscopy and Nondestructive Evaluation

Noriyoshi Chubachi, Cochairman
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Invited Papers

8:00

X1. A study of the mechanical property of materials using acoustic microscopy. Kazushi Yamanaka (Mechanical Engineering Laboratory, 1-2 Namiki, Tskuba, 305 Japan)

The interpretation of the general features of acoustic microscopy images is discussed in conjunction with its application to materials science under the following topics. (1) Interference fringes have been discovered around surface breaking cracks, and a model based on the leaky Rayleigh wave has been proposed. It has attracted a considerable amount of interest because these fringes might give new information on the elastic properties of microscopic discontinuity such as cracks, grain boundaries, and steps that are almost normal in sample surfaces. (2) When subsurface cracks are close to parallel to the sample surface, they can be analyzed by a simple plate model with water or air below. Either the $V(z)$ curve measurement or the resonance method is useful depending on whether the frequency-depth product is much smaller or close to the first cutoff of the higher-order modes of the leaky Lamb waves. This analysis was successfully applied to a study of indentation fractures of steel ball bearings in use. (3) Image observation and the leaky SSCW velocity measurement at 775 MHz have been applied to individual spherulites in semicrystalline polypropylene. Through the signal processing of the $V(x,z)$ image, it has been found that the difference in velocity between spherulites with different thermal histories can be larger than 30%.

8:20

X2. Review of scanning laser acoustic microscopy—Imaging applications. Lawrence W. Kessler (Sonoscan, Inc., 530 East Green Street, Bensenville, IL 60106)

The scanning laser acoustic microscope (SLAM) was first developed in the early 1970s as a laboratory curiosity to make high-resolution images of materials in the expectation that microstructure could be studied in a new way. Since that time the techniques have developed into practical and commercially available instruments that have found their way into industry by solving very important problems in quality control. Stimulated by the evolution of new materials and processes, for example, fine ceramics, composites, and the ever-changing field of microelectronics, SLAM has often played an important role as an analytical tool, alongside other analytical instruments to "see inside" products nondestructively, in true real-time and with higher resolution than had been possible with conventional ultrasonic imaging.

8:40

X3. Recent progress in quantitative measurements by acoustic microscopy. Jun-ichi Kushibiki (Department of Electrical Engineering, Tohoku University, Sendai, 980 Japan)

The line-focus-beam (LFB) acoustic microscopy developed at Tohoku University in 1981 is expected to provide a new technology for quantitative material analyses in scientific and industrial fields. The material

characterization method through the so-called $V(z)$ curve analysis can measure the acoustic properties of materials, viz., velocity and attenuation, of leaky surface acoustic waves (LSAWs) excited on the boundary between samples and its reference coupling liquid of distilled water. The most remarkable features of the method are the following: the proper anisotropy detection of the material properties, very high measurement accuracy, and nondestructive and noncontracting evaluation. In this paper, some of the promising applications of the acoustic microscopy will be reviewed concerning the following subjects: the characterization of the acoustic anisotropy and inhomogeneity of materials; the structural analysis of polycrystalline materials; measurement of film thickness; and determination of the elastic constants of bulk and thin-film materials. Present and future problems will also be discussed.

9:00

X4. Quantitative measurement capabilities of the scanning laser acoustic microscope. William D. O'Brien, Jr. (Bioacoustics Research Laboratory, Department of Electrical and Computer Engineering, University of Illinois, 1406 West Green Street, Urbana, IL 61801)

The scanning laser acoustic microscope (SLAM) has the capability of determining the spatial distributions of attenuation coefficient and speed in materials under evaluation. The attenuation coefficient is determined by the insertion loss procedure [K. M. U. Tervola *et al.*, IEEE Trans. Sonics Ultrason. **SU-32**, 259–265 (1985)], which compares the received signal amplitude of a specimen of known thickness in the sound path with that of the reference medium. The entire image is broken into 64 subimage areas, each of approximately 400 μm $\times$ 250 μm in size. This subarea image is used to determine the insertion loss. The slope of the linear least-squares fit line for an insertion loss versus specimen thickness plot yields the attenuation coefficient. Using known materials, the attenuation coefficient accuracy and precision are $\pm$ 12% and $\pm$ 15%, respectively. Ultrasonic speed is determined from the interference mode image by way of the spatial frequency domain technique [K. M. U. Tervola and W. D. O'Brien, Jr., IEEE Trans. Sonics Ultrason. **SU-32**, 544–554 (1985)]. The image's field of view (3 mm $\times$ 2 mm) contains approximately 39 vertical interference lines equally spaced about 85 μm apart. The speed of sound is determined by the horizontal shift of the interference lines between the reference medium and the specimen. Again, using a known material, the speed accuracy, and precision are $\pm$ 2.9% and $\pm$ 0.4%, respectively. Smaller regional differences can be distinguished with the speed analysis than the attenuation coefficient analysis. Each speed pixel is approximately 4 μm $\times$ 8 μm and a speed profile (80 μm $\times$ 2 mm) along the vertical direction of the image is generated. In contrast, in each insertion loss pixel (400 μm $\times$ 250 μm), an average insertion loss value per section is generated over a region approximately 1 mm $\times$ 1 mm in extent. Further, at least three separate sections are required to calculate the attenuation coefficient. The application of SLAM towards an understanding of ultrasonic interaction action with biological materials is being studied by a number of different animal and tissue models. These models consider both normal and pathologic, homogeneous and heterogeneous, and soft and hard tissues. [Work supported by NIH Grants AM21557, CA30629, and AR36794.]

9:20

X5. Recent developments in ultrasonic nondestructive evaluation in Japan. Hidekazu Fukuoka (Faculty of Engineering Science, Osaka University, Toyonaka, 560 Japan)

Ultrasonic nondestructive evaluation is one of the most powerful tools among many nondestructive evaluation procedures and plays an important role in the on-line manufacturing process and postprocess material testing. Recent developments in this field in Japan at universities, government laboratories, and industrial laboratories are surveyed. The behavior of ultrasonic waves in solids is vividly understood by both computer simulation and stroboscopic photoelastic visualization and it provides a fundamental basis of quantitative information for sizing flaws. The acoustoelastic measurement of residual stress in structural components allows us to estimate their service-life expectancy. By the acoustoelastic evaluation of texture in steel, an ultrasonic pole figure is obtained more conveniently than with the x-ray method. As industrial applications, there are the laser generation and detection of ultrasonic waves, the detection of surface defects and internal voids in ceramics, phased array transducers and their application for steel pipe testing, and on-line wall thickness measurements using an electromagnetic acoustic transducer for hot seamless steel pipes. The acoustic microscope also has many applications in the nondestructive evaluation of materials.

9:40

X6. Characterization of surface-breaking and near-surface cracks by the acoustic microscope. J. D. Achenbach (Center for Quality Engineering and Failure Prevention, Northwestern University, Evanston, IL 60208)

The acoustic microscope is now becoming a more frequently used instrument for the study of surfaces and near-surface inhomogeneities. One potentially important application concerns the detection and characterization of very small surface-breaking or subsurface cracks. For this application, it is important that the instrument can be used in the quantitative rather than in the imaging mode. It is expected that surface waves in conjunction with a line-focus configuration would provide the most effective probing modality. An effective use of this modality will, however, require insight in the interaction of surface waves with cracks. It is the objective of this presentation to provide such insight on the basis of analytical and numerical solutions to the interaction problems. In this work, the crack is inclined under an arbitrary angle with the normal to the free surface of a half-space. The representation integral for the scattered displacement field is employed to obtain expressions for the surface tractions. In conjunction with the conditions on the crack faces and on the surface of the half-space, these expressions yield a set of boundary integral equations, which are solved numerically by the boundary element method. Results for the reflection and transmission coefficients, as well as for the time-averaged energy-flux radiated onto the solid by body waves, are presented. Parametrical studies reveal the variation of these results with the angle of inclination of the crack, its length, and with the frequency of the incident wave.

10:00

X7. Nondestructive measurement of texture and nine elastic constants of rolled metal sheets by EMAT. Katsuhiro Kawashima (Electronics and Control Systems Laboratory, R&D Laboratories-I, Nippon Steel Corporation, Nakahara-ku, Kawasaki, 211 Japan)

All the lowest-order independent coefficients W400, W420, and W440 for the crystallographic orientation distribution function were obtained nondestructively for thin rolled steel sheets by measuring ultrasonic wave speeds and making use of the known values of the single-crystal elastic constants. All the ultrasonic wave speeds were measured by EMAT so that no acoustic coupling medium was necessary, and the measurements were done quickly and with very good reproducibility. Pole figures were calculated using the obtained coefficients W400, W420, and W440 and the results were quite similar to the pole figures obtained by the x-ray diffraction method. It is well known that a rolled steel sheet can be modeled as an orthorhombic continuum having nine different elastic constants. It is also known that only six of the nine constants are independent when the sheet is a polycrystal of cubic crystallites and assuming that the macroscopic properties can be predicted by an averaging scheme. All these nine constants were calculated using the obtained W400, W420, W440, and the known single-crystal elastic constants. Young's moduli in the plane of the rolled steel sheet were also calculated and compared with the measured plastic strain ratio. Good relations were obtained between the average Young's modulus E and the average plastic strain ratio r.

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10:20

X8. A study of generalized Lamb waves in solid-state bonded aluminum-steel samples. Laszlo Adler,[a] Michel deBilly, Gerard Quentin (Groupe de Physique des Solides, Université Paris 7, Paris, France), Maryline Talmant,[b] and Peter B. Nagy (Department of Welding Engineering, The Ohio State University, Columbus, OH 43210)

Wave propagation in the aluminum layer on a steel substrate and the steel layer on an aluminum substrate has been studied. These material combinations correspond to the special case where the transverse velocities are close to each other while the densities and longitudinal velocities are not. Experimental measurements were carried out using two different methods: (i) single transducer broadband, and (ii) double transducer narrow band. Experimentally obtained dispersion curves for the phase velocities were compared to existing theoretical calculations. A method to evaluate the quality of solid-state bond based on the measured dispersion curve is also suggested. [Work supported by D.O.E.]. [a] Permanent address: Department of Welding Engineering, The Ohio State University, Columbus, OH 43210. [b] Permanent address: Groupe de Physique des Solides, Université Paris 7, Paris, France.

Contributed Papers

10:40

X9. Electromagnetically generated acoustic determination of delamination. Wayne Imaino (IBM Research Laboratory, Almaden Research Center, K08/282, 650 Harry Road, San Jose, CA 95120-6099)

Previous work has demonstrated a technique for acoustically detecting localized delamination of metallized patterns on insulating substrates [W. Imaino *et al.*, in *Proceedings of the IEEE Ultrasonics Symposium*, edited by B. R. McAvoy (IEEE, New York, 1986), p. 1065]. Employing a high-spatial resolution permeable EMAT, to preferentially excite the metallization, the acoustic coupling between the metal foil and substrate may be probed. Scanning, then provides a map of delaminations. To extend and generalize these results, the detailed generation mechanism and acoustic response of the substrate has been studied. A computer program developed previously, [W. Imaino, J. Acoust. Soc. Am. Suppl. 1 **80**, S7 (1986)] is used to model the acoustic source and a finite element calculation provides a detailed description of the acoustic behavior. The effect of source size relative to defect dimensions and acoustic properties of the substrate have been studied. The determination of the localized coupling is complicated by structural resonances of the substrate, which make single frequency measurements unfavorable. However, the analysis shows that the spectral acoustic behavior provides an indication of metallization to substrate coupling. A signal processing algorithm based on this analysis has been formulated and will be described.

10:52

X10. Abstract withdrawn.

11:04

X11. Application of ultrasonic computed tomography to nondestructive inspection of SiO_2 ingot material qualities. Hiroaki Yamada (College of Industrial Technology, Nihon University, Narashino, 275 Japan), Yoshiro Tomikawa, and Mitsuhiro Nishida (Department of Electrical Engineering, Yamagata University, Yonezawa, 992 Japan)

It is very important industrially to produce SiO_2 chips with uniform material qualities for integrated circuit elements (IC). Therefore, nondestructive inspection to detect whether the material quality of the SiO_2 column ingot is uniform or not is necessary before it is cut into a small chip for IC elements. For this purpose, the x-ray CT technique is effective. However, the technique is dangerous for workers in usual factories. Therefore, ultrasonic CT was tried for such detection. The ultrasonic CT was already developed using data from the time of flight (TOF) and applied to certain cases; for example, nondestructive inspection of rotten parts or hidden knobs in a cedar log and the measurement of internal temperature distribution. In this paper, an attept was made to use the ultrasonic CT for the nondestructive inspection of SiO_2 column ingots. The experiment was made with an ultrasonic transducer at 5.0 MHz. The results show the usefulness of the TOF ultrasonic CT for the nondestructive inspection of SiO_2 column ingots.

11:16

X12. Ultrasonic properties in excessively fatty rat livers. William D. O'Brien, Jr. (Bioacoustics Research Laboratory, Department of Electrical and Computer Engineering, University of Illinois, 1406 West Green Street, Ubana, IL 61801), James F. Greenleaf (Mayo Clinic, Rochester, MN 55905), and John W. Erdman, Jr. (Department of Food Science, University of Illinois, Urbana, IL 61801)

The orotic acid dietary "liver lipid" model has successfully been used to study ultrasonic properties in excessively fatty rat livers [O'Brien *et al.*, J. Acoust. Soc. Am. **83**, 1159–1166 (1988)]. The model was used again in order to evaluate more fully ultrasound properties within a more condensed (2%–10%) lipid concentration range. The fatty livers were first evaluated in the intact animal (*in situ*) with an ultrasonic imaging system ($\simeq 5$ MHz) and then the excised (*in vitro*) livers were evaluated over the 1- to 100-MHz frequency range. Water, lipid, total protein, and collagen were evaluated for each liver. Significant ($p < 0.001$) correlations of ultrasonic properties (absorption, all attenuation, speed, and Markovian entropy) against tissue properties (water and lipid) were obtained, thus suggesting that the independent measure of texture may relate to basic ultrasonic propagation properties. [Work supported by NIH Grant CS36029.]

11:28

X13. Acoustic microscope with an acousto-optic array detector. Bill D. Cook, J. Ted Miller (Department of Mechanical Engineering, University of Houston, Houston, TX 77204-4792), and D. Kent Lewis (Lawrence Livermore National Laboratory, Nondestructive Evaluation Section L-333, Livermore, CA 94550)

This very low-frequency microscope is a pitch–catch device where the receiver is a large, synthetic, linear acousto-optic array. The focused ultrasound is transmitted through and reflected from a sample off-axis. The reflected or transmitted pressure is sampled along several lines with a laser and photodiode. This recorded information is then processed to find the angular distribution of the sound. Comparison with incoming sound yields reflection and transmission coefficients. Measurements have been made on thin plates with both pulse and continuous waves. With continuous waves, the reflection and transmission coefficients are sometimes greater than unity, probably due to mode conversion. Pulsed waves yield presentations of dispersion of the modal velocities. [Work supported by NATO and DOE.]

11:40

X14. Scattering-induced error in longitudinal wave velocity measurements. Hiroki Tada, Hidekazu Fukuoka, and Tomohiro Yamasaki (Faculty of Engineering Science, Osaka University, Toyonaka, 560 Japan)

The scattering of an elastic longitudinal wave on reflection at a free-plane boundary is studied using the analytical solution for a two-dimensional wave field generated by strip surface stresses. The variation in the scattered field is calculated with various incident angles of the narrow beams and displayed in polar diagrams, which show a wide range of scattered strength. The calculated result reduces to the specular reflection of an infinite plane wave with an increase in the number of beams in the bundle. These results give the basis for estimating the experimental error due to the scattering of the first and second echoes launched by transducers of finite width. The scattering-induced error is well characterized by the ratio of the transducer width to the square root of the specimen thickness, both being normalized by the wavelength. As this ratio increases from zero, the error begins to increase, passes a peak, and then approaches a constant value. To minimize the error, the use of a transducer is suggested for which the nearfield length is at least four times the specimen thickness.

11:52

X15. Low-frequency search unit for ultrasonic testing of concrete. Kanji Imoto (Department of Engineering, Meiji University, 1-1-1 Higashimita, Tama-ku, Kawasaki, 214 Japan), Koji Ohta (Nikki Inspection Services Co., Ltd., 3-35 Onoe-cho, Naka-ku, Yokohama, 231 Japan), and Takeharu Watanabe (JGC Corporation, 1-14-1 Bessho, Minami-ku, Yokohama, 232 Japan)

A new low-frequency search unit for ultrasonic testing has been developed. A conventional low-frequency search unit involves many high-harmonic components in its signal. This is one of the reasons why it is difficult to test concrete by the ultrasonic pulse echo method. Characteristics of a search unit have been analyzed by a computer. This analysis was based on the assumption that each of the components of the search unit, such as the piezoelectric transducer and the resonator will form a four-port terminal network. The design criteria for a search unit have been determined by this analysis. The model for analyzing the search unit characteristics and the results of analysis are shown. A low-frequency search unit has also been produced. The characteristics of this search unit are compared with analytical results. Ultrasonic testing of concrete has been improved by using this search unit.

Session Y. Physiological Acoustics IV and Psychological Acoustics I: Auditory Prostheses; Hearing Impairment; Auditory-Evoked Potentials (Poster Session)

Charles S. Watson, Cochairman
Department of Speech and Hearing Sciences
Indiana University
Bloomington, Indiana 47405

Keiichi Murata, Cochairman
Department of Neurophysiology
Medical Research Institute
Tokyo Medical and Dental University
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Contributed Papers

All posters will be displayed from 8:00 a.m. to 12:00 noon. To allow contributors the opportunity to see other posters, contributors of papers Y1 through Y9 will be at their posters from 8:00 to 9:30 a.m. and contributors of papers Y10 through Y21 will be at their posters from 9:30 to 11:00 a.m.

Y1. High-fidelity hearing aid amplifier. Mead C. Killion (Etymotic Research, 61 Martin Lane, Elk Grove Village, IL 60007)

If the claims of many mildly hearing-impaired individuals that they "do not need a hearing aid most of the time" are true, the appropriate hearing aid design appears self-evident: the hearing aid should be designed to do nothing (be acoustically transparent) most of the time. A low-power-consumption bipolar linear (!) custom integrated circuit hearing aid amplifier has been designed to permit that result while providing the gain and high-frequency emphasis usually required to make quiet sounds audible. The goal is to amplify only the quiet sounds the wearer is missing, while neither standing in the way of his apparently adequate hearing for high-level sounds, nor—worse yet—amplifying sounds that are already loud enough. The resulting level-dependent frequency response characteristic is opposite that of the presently popular "ASP" circuits, which appear to be solving a problem *created* by the typical hearing aid design. [Work supported in part by NIA.]

Y2. Experimental assessment of a method for calibration of ear canals at high frequencies. Robert Berkovitz and Kenneth N. Stevens (Sensimetrics Corporation, 1 Kendall Square, Cambridge, MA 02139)

An earlier publication [Stevens *et al.*, J. Acoust. Soc. Am. **81**, 470–484 (1987)] described a procedure for determining the absolute sound pressure at the medial end of the ear canal at frequencies in the range 8–20 kHz when a sound source is coupled to the ear through a lossy tube. The present paper reports comparisons of estimated and measured apex sound pressures for various ear-canal models. Agreement was within 1–2 dB for simplified models with rigid perpendicular and oblique terminations. Changes in length over a range similar to that of human ear canals had little effect on results. Similar agreement up to about 18 kHz was obtained for nonuniform ear-canal models with constrictions to less than one-half the cross-sectional area of the average ear canal, and the error was slightly larger above 18 kHz. Constrictions in the models led to nonuniformities in the frequency response at the apex. Estimates of apex sound pressures for a number of human ears exhibited features similar to those observed for the models. Improvements in signal processing and fitting techniques made since the earlier publication are described. [Work supported by a grant from NINCDS. K. N. Stevens is also at the Massachusetts Institute of Technology.]

Y3. Development of hearing aids without earphones. H. Ono (Department of Audiology, RIEEC, Tokyo Gakugei University, Koganei, 184 Japan) and J. Kanzaki (Department of O.R.L., School of Medicine, Keio University, Tokyo, 160 Japan)

Inserting earphones in your ear canal can be uncomfortable sometimes. In fact, many people with light and moderate hearing impediments do not wear earphones for this reason, although they know that hearing aids can greatly enhance their hearing. To solve this problem, a new system of hearing aid was developed that will make sounds sound louder in certain areas—around the subject's ears—than in other areas of the room. Three methods for the system were devised. They use: (1) interference of ultrasonic interference, (2) nonlinear principle of parametric speakers, and (3) equipment to accumulate sound reflection. For each method, studies were conducted on: the S/N ratio of output sound pressure to sounds in other areas, energy conversion efficiency, methods to cancel outside noise, effect on human health, and the size and price of the device. All three methods were found to be feasible. Units were built on a trial basis using the three above methods, and are being studied to see if they can actually be used as hearing aids.

Y4. Interaction between compression limiting and frequency gain characteristics. H. Cynthia Link, Edward A. Cudahy (Lexington Center, 30th Avenue and 75th Street, Jackson Heights, NY 11370), and Harry Levitt (City University of New York—Graduate Center, 33 West 42nd Street, New York, NY 10036)

The interaction between frequency gain characteristics and compression limiting was investigated using an adaptive paired comparison strategy to estimate the optimum post-limiter frequency response for each of three pre-limiter frequency gain characteristics. Frequency-dependent compression limiting paralleling the subject's loudness discomfort level (LDL) curve was used. Stimulus intensity was adjusted to provide three levels of compression limiting. Results for both preference and intelligibility judgments with continuous discourse in quiet showed that most subjects tended to compensate for manipulations of the pre-compression filters with adjustments in the post-compression filters. The amount of compression had only a secondary effect on filter selection. [Research supported by NIDRR.]

Y5. Masked thresholds for hearing-impaired listeners. C. Formby (Department of Communicative Disorders, JHMHC/Box J-174 and Psychoacoustics Laboratory, Department of Psychology, University of Florida, Gainesville, FL 32610)

Masked thresholds for sinusoids between 1000–6500 Hz in (1) broadband noise, (2) low-pass filtered noise, and (3) high-pass filtered noise were measured by adaptive forced-choice procedure for hearing-impaired listeners [acute Meniere disease ($n = 3$), noise-induced hearing loss ($n = 7$), and eighth-nerve tumor ($n = 2$)]. Thesholds in the filtered noises were measured as a function of masker cutoff frequency. Masker spectrum levels (55–70 dB SPL) were listener dependent and were equal for the three masker conditions. Masked thresholds for the broadband masker were near normal except for the tumor patients, who showed abnormally large (by 10–15 dB) masked thresholds at 1000 Hz. Auditory filters were estimated by plotting the masked thresholds for the low-pass and high-pass marker conditions at the marker cutoff frequencies. Linear functions were fitted to the masked thresholds to derive the auditory filter's half-power frequencies and bandwidth. The auditory filters were characterized by shifts in the center frequency of the filter below signal frequency and, often, by broad tuning. [Work supported by NIH.]

Y6. Binaural detection and discrimination for listeners with high-frequency sensorineural hearing losses. J. Koehnke, H. S. Colburn, and G. A. Owen (Boston University, 48 Cummington Street, Boston, MA 02215 and Research Laboratory of Electronics, MIT, Cambridge, MA 02139)

As part of a survey study of binaural performance, NoSπ detection thresholds and just-noticeable differences (jnd's) in interaural time delay (ITD) and interaural intensity difference (IID) were measured for listeners with high-frequency sensorineural hearing losses. Thresholds and jnd's were obtained with 1/3-oct noise bursts at 500 and 4000 Hz for a variety of interaural reference conditions, including ITDs and IIDs up to 600 μs and 46 dB. The present data for high-frequency hearing loss subjects show poorer than normal binaural performance even at 500 Hz, a region where auditory thresholds are normal; however, binaural detection and discrimination performance at 4000 Hz is generally poorer than at 500 Hz. This is in agreement with previously reported results [e.g., J. Koehnke and H. S. Colburn, J. Acoust. Soc. Am. Suppl. 1 **81**, S27 (1987)]. In cases with unilateral losses studied to date, the compensation for interaural threshold differences through appropriate choice of the reference interaural intensity differences does not improve binaural performance. [Work supported by NIH.]

Y7. Amplitude compression for cochlear implant speech processors. R. Benjamin Knapp, Laurel J. Dent, David H. Liang (Stanford Electronics Laboratories, AEL 132, Stanford, CA 94305), and Robert V. Shannon (Boys Town National Institute, 555 North 30th Street, Omaha, NE 68131)

One requirement in speech processor design for cochlear implants is determining the instantaneous mapping of the broad acoustic dynamic range into the narrow dynamic range for electrical stimulation. Magnitude estimation was used to measure implanted subjects' growth of loudness for several stimulus waveforms including sinusoidal, pulsatile, and amplitude modulated pulsatile signals. Loudness growth was best modeled by a power law function of the form first proposed by Stevens [J. Acoust. Soc. Am. **27**, 815–829 (1955)] to describe normal listeners' loudness growth for acoustic stimuli. The power function exponent varied with electrical stimulus type, but when loudness growth was expressed as a function of proportion of dynamic range, a single exponent described all the observations. The Stanford real-time speech processor was programed to permit the subject to adjust a fractional power compression designed to elicit normal loudness growth by correcting for the difference between the normals' and implant subjects' exponents. The "correction exponent" was confirmed to be the same for all electrodes and waveforms. The physiological implications of the findings and the effects of the compression on speech discrimination will be discussed.

Y8. Error analysis of cochlear implant consonant distributions. Gerald S. Wasserman (Department of Psychological Sciences, Purdue University, West Lafayette, IN 47907) and Richard T. Miyamoto (Department of Otolaryngology, Indiana University School of Medicine, Indianapolis, IN 46202)

Consonant discrimination in a two-alternative, forced-choice design was measured in adult- and juvenile-onset cochlear implant patients under control (3M-House) and receptor-coded conditions [cf. J. Acoust. Soc. Am. Suppl. 1 **82**, S71 (1987)]. Errors were not symmetric but dependent on which alternative of a pair of consonants had been spoken. Errors conserved phonemic identity for adult- but not for juvenile-onset patients. A two-dimensional (control versus receptor coding) coincidence filtration of adult-onset errors implicated second-degree regulation of performance by the various articulatory features of speech. This implication was tested with a regression analysis: it demonstrated a reliable influence of certain articulatory features, particularly of voicing and plosion, in both conditions. A coding-dependent shift of articulation influence was found: Voicing became a more reliable cue with receptor coding, manner cues changed subtly between conditions, and place became a reliable cue with receptor coding. [Work supported by NSF.]

Y9. Temporal and spatial interaction of pulsatile electrical stimuli of the auditory nerve in cats. David H. Liang (AEL 128, Department of Electrical Engineering, Stanford University, Stanford, CA 94305)

It is fairly clear that the separate electrodes of a cochlear prosthesis do not represent independent channels for transmitting information. The currents from the different electrodes interact both spatially and temporally. To examine these interactions, recordings were made from single units in the VIIIth nerve of cats that had an eight-channel stimulating array acutely implanted in the scalar tympani. Spatial interaction was studied by determining the threshold for simultaneous in-phase stimulation of two electrodes. It was found that the resulting thresholds were not always predicted well by simple linear summation of the effectiveness of the two electrodes used in isolation. Temporal interaction was examined by studying the effect of a subthreshold stimulus on the threshold of a subsequent stimulus. This experiment has particular importance for stimulation schemes that seek to reduce interaction between adjacent electrodes by interleaving them in time. The results suggest that, for biphasic stimulation, the residual effect of stimuli up to 75% of threshold is minimal. [Work supported by NIH.]

Y10. Distortion-product emissions in humans with high-frequency sensorineural hearing loss. Frances P. Harris and Theodore J. Glattke (Department of Speech and Hearing Sciences, University of Arizona, Tucson, AZ 85721)

Distortion-product emissions (DPEs) were detected in 20 adults with pure-tone behavioral threshold elevation at frequencies above 1000 Hz and in 20 adults with normal hearing. Equilevel pure tones at $f1$ and $f2$ ($f2/f1$ approximately 1.21) ranged from 65 to 20 dB SPL and were located at frequencies that approximated audiometric frequencies from 750 to 8000 Hz. DPEs at $2f1 - f2$ ranged in amplitude from 23 to -18 dB SPL and were from 30 to 60 dB below the primaries for subjects with normal hearing. DPE amplitudes, response ranges, and slope values were significantly different between the two groups when the primaries were in the high-frequency region, but were similar at 0.75 and 1 kHz. DPEs were present for frequencies at which pure-tone behavioral thresholds were less than 15 dB HL and were absent or significantly attenuated if threshold exceeded 50 dB HL. The amplitude and range of DPE responses varied considerably when behavioral thresholds were between 20 and 45 dB HL. Results are compatible with similar findings from experimental animals.

Y11. Acoustic and psychoacoustic evaluation of suppression and synchronization of spontaneous otoacoustic emissions. Glenis R. Long, Savithri Sivaramakrishnan (Department of Audiology and Speech Sciences, Purdue University, West Lafayette, IN 47907), and Arnold Tubis (Department of Physics, Purdue University, West Lafayette, IN 47907)

The impact of external tones on spontaneous otoacoustic emissions (tonelike stimuli of cochlear origin) depends on both the level and relative frequency of such tones. The most commonly measured kinds of interaction are suppression, beating, and entrainment. Psychophysical correlates of these forms of interaction are masking, monaural diplacusis, and some aspects of threshold microstructure. The relation between the different acoustic and psychophysical observations is investigated in three individuals with large spontaneous emissions both under normal circumstances and when the emissions are reduced by aspirin consumption. The acoustic observations will be compared to the output of digital cochlear models which assume that an otoacoustic emission can be modeled as a limit-cycle oscillator.

Y12. Characteristics of electrically induced otoacoustic emission. Toshio Moriyama, Yutaka Hosokawa, Sadao Minami, and Keiichi Murata (Department of Neurophysiology, Medical Research Institute, Tokyo Medical and Dental University, Chiyoda-ku, Tokyo, 101 Japan)

Acoustic emissions were detected using the fast Fourier analysis of sounds in an ear canal during injection of alternating currents at different frequencies into the scala media of guinea pigs anaesthetized with urethane (1.2 g/kg i.p.). The amplitude of the distortion products relative to the fundamentals in the emitted sound was about ten times as large as that in the cochlear current, which was already slightly distorted, probably due to nonlinearity at the metal electrode surface. This discrepancy implies a larger additional nonlinearity in the cochlea. During COCB stimulation or after moderately intense sound exposure, the emitted distortion products were suppressed reversibly, in contrast to almost no change in the fundamental emissions. All the emissions were suppressed reversibly after temporary anoxia or furosemide administration, and they disappeared completely after severe acoustic trauma or death. These results suggest that the normal metabolic activity of the cochlea is required for generation of the emissions, and that the electromechanical transduction is less vulnerable than the cochlear nonlinearity.

Y13. Longitudinal analysis of spontaneous otoacoustic emissions in audiometrically normal adults. Holly Susan Haggerty (Division of Otolaryngology, R-135, Stanford University School of Medicine, Stanford, CA 94305)

The second-by-second frequency stability of a spontaneous otoacoustic emission is purportedly high [W. Bialek and H. P. Wit, Phys. Lett. A **104**, 173–178 (1984)], although minute-by-minute recordings have shown less stability. Additionally, while spontaneous emissions have been described as stable over years in a few subjects, diurnal and monthly variations have been reported for a few others [W. Fritze, Arch. Otorhinolaryngol. **238**, 189–196 (1983); H. P. Wit, Hear. Res. **18**, 197–199 (1985)]. To partially clarify this issue, spontaneous emissions were recorded from 11 normal-hearing adults. Diurnal, weekly, and monthly stability and variation of frequency amplitude were analyzed over the course of 2 months. Preliminary results indicate impressive intrasubject frequency and intensity variations. Final results are discussed with respect to theoretical modeling of active processes in the cochlea. [Work supported by the Knowles Research Foundation.]

Y14. Spontaneous otoacoustic emissions cause disruptive tinnitus in 4 of 88 subjects. M. J. Penner (Department of Psychology, University of Maryland, College Park, MD 20742)

Experimental tests linking SOAEs with problem tinnitus are detailed by presenting one case study of a female subject whose tinnitus consists of an annoying SOAE in the right ear and tinnitus unrelated to SOAEs in the left ear. The tests involve demonstrating that the tinnitus only disappeared when a pure tone suppressed the SOAE and another continuous tone masked the tinnitus, and that the isomasking contour for the tinnitus caused by the SOAE was frequency specific whereas the contour for the tinnitus unrelated to the SOAE was not. Using similar tests on 87 additional subjects, an estimate was made of the percent of tinnitus sufferers whose tinnitus was caused by SOAEs: the 95% confidence limits of this estimate were 0.20% and 9%. [Work supported by NIH.]

Y15. Fundamental observations for tinnitus. Teiji Tanahashi, Koichi Tsuzuki, and Atsushi Tanahashi (Department of Otolaryngology, Nagoya University Branch Hospital, 1-1-20 Daikominami, Higashi-ku, Nagoya, 461 Japan)

Changes in the spontaneous discharges of the auditory neuron in the cochlear nucleus were analyzed under various conditions. Young adult cats were anesthetized with pentobarbital, pentazocine, and diaxepam. Characteristic frequency, threshold, PST histogram, and the spontaneous discharge rate of each neuron were obtained before and following asphyxia. EEG, ECG, and blood gas content were also analyzed. Following asphyxia, lasting from 60–110 s, the spontaneous discharges increased markedly in ten neurons, but in five neurons, the increase was slight or nonexistent. The spontaneous discharge of the auditory neuron in the cochlear nucleus following asphyxia was temporarily suppressed by the administration of Lidocaine or Diltiazem. This effect was observed continuously for over 3 h. The dose effect was also examined.

Y16. Predictors of TTS after pure-tone exposure in the cat. Michael L. Wiederhold, J. Luis De La Cruz, Louis Metzman, and Thomas J. Prihoda (Division of Otorhinolaryngology and Department of Pathology, University of Texas Health Science Center and Audie L. Murphy V. A. Hospital, San Antonio, TX 78284-7777)

Anesthetized cats were exposed to 4-kHz pure tones, of varying level and duration. The N_1 auditory-nerve responses to filtered clicks, at 2, 4, 6, 8, 10, and 15 kHz, as well as $2f_1 - f_2$ distortion product (DP) signals in the ear canal, were recorded before and after exposure. The f_1, f_2 pairs were 4.0 and 5.2 kHz for all animals and, additionally, 1.6 and 2.1 kHz and 4.6 and 6.0 kHz in a subset of animals. In all cases, $L_1 = L_2 + 10$ dB. Multiple polynomial regression analysis, relating pre-exposure 10-μV N_1 thresholds at 4 and 6 kHz, $-$ 10-dB SPL DP thresholds for 4.0- and 5.2- and 4.6- and 6.0-kHz primaries and noise dose (total energy) to 6-kHz N_1 threshold shift (TS) and DP TS (4.0- and 5.2-kHz primaries), as outcomes. The TS for both N_1's and DPs displayed considerable variability between animals, even with the same exposure. Pre-exposure 4- and 6-kHz thresholds were the best predictors of 6-kHz TS ($r^2 = 0.57$ and 0.58, respectively, $p = 0.0001$) and 6-kHz TS was the outcome that could be predicted with the most certainty. Addition of DP threshold did slightly increase the predictive ability of the model. Pre-exposure N_1 threshold only weakly predicted DP TS. The DP TS assessed 6-kHz N_1 TS weakly ($r^2 = 0.27$) but significantly ($p = 0.014$). The poorer ability of DP threshold and DP TS to predict N_1 TS, compared to pre-exposure N_1 thresholds, is consistent with the variability between animals in DP levels, which are susceptible to abnormalities in forward conduction, sensory function, and reverse conduction of DP signals into the ear canal. [Work supported by Veterans Administration Medical Research Funds.]

Y17. Effects of high-level impulse noise intensity, number, and rate on hearing. Roger P. Hamernik and William A. Ahroon (Auditory Research Laboratory, State University of New York, Plattsburgh, NY 12901)

The effects of very low-frequency, energy-content impulse noise exposures on hearing were studied in a population of 109 chinchillas exposed to 1, 10, or 100 impulses presented at 150-, 155-, or 160-dB peak SPL at rates of one every 6, 60, or 600 s. The TTS and PTS were measured using the auditory evoked potential. Histological measures were obtained using standard surface preparation techniques. The following preliminary conclusions can be made. (1) There was no statistical difference in the amount of hearing loss or the amount of sensory cell loss for exposure to a single impulse at 150 or 155 or 160 dB peak SPL. Individual animals showed no permanent hearing loss and no significant sensory cell loss. (2) There was a considerable amount of variability or degree of susceptibility across animals as the severity of the exposure increased. This increase in susceptibility seemed to be tied more to peak levels of the stimulus than to the total energy. The variability produces, in extreme cases, a complete dichotomy in the results. (3) The permanent effects seemed to be dependent upon peak levels more than upon the total energy in the exposure stimulus. Also, for a constant peak and energy level, the more rapid presentation rate produced the greatest effect.

Y18. Effects of periodic rest on hearing loss and cochlear damage following interrupted exposure to high-frequency noise. W. W. Clark, B. A. Bohne, and T. J. Reilly (Central Institute for the Deaf, 818 South Euclid Avenue, St. Louis, MO 63110)

Two groups of chinchillas were exposed to an octave band of noise (OBN) centered at 4.0 kHz, 86 dB SPL for 6 h per day. The first, made monaural prior to the start of the exposure, was exposed for 36 days. The second was made monaural after 36 days of exposure and exposed for an

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additional 36 days. Thresholds were assessed twice daily with a positive reinforcement technique, and the cochleas were prepared for microscopic examination 3 months after the exposure. All subjects sustained initial threshold shifts of 45–55 dB at test frequencies within the exposure band (3.35–8.0 kHz); after 5–10 days, threshold shifts began to decline and recovered about 20 dB over the next 10 days of exposure. Thresholds were stable after about 20 days of exposure for both the 36- and 72-day groups. Small, permanent threshold shifts were measured within the exposure band for both groups of animals. Cochleas exposed for 36 days had mild but significant damage in the basal turn; those exposed for 72 days sustained slightly more damage. [Work supported by Grant No. 0H 02128 from NIOSH.]

Y19. Relations between frequency selectivity, temporal resolution, and speech intelligibility in sensorineural hearing-impaired subjects. Kiyoshi Yonemoto, Noriko Kurauchi (National Rehabilitation Center for the Disabled, 4-1 Namiki, Tokorozawa, 359 Japan), Hareo Hamada, and Tanetoshi Miura (Tokyo Denki University, 2-2 Kanda-Nishiki-cho, Chiyoda-ku, Tokyo, 101 Japan)

Frequency selectivity (the shape of the auditory filter), temporal resolution (the ability of gap detection), and speech intelligibility were measured for hearing-impaired subjects, and the relationships among these were investigated. The subjects were aged 18–49 years and had sensorineural hearing loss. Their average hearing threshold level was above 70 dB HTL. The shape of the auditory filter was measured from pure-tone threshold values under various maskers of notched broadband noise. Gap detection thresholds were measured using octave-band noise as a stimulus. A speech intelligibility test was conducted using two-syllable female speech. The results showed no correlation between hearing threshold level and any one of the following three parameters: auditory filter shape, gap detection threshold, and speech intelligibility score. However, the present subjects with severe hearing loss also showed a strong correlation between the sharpness of auditory filter and the speech intelligibility score, and a weak correlation between the gap detection threshold and the speech intelligibility score.

Y20. Evaluation of auditory enhancement effect in normal and hearing-impaired listeners. Linda M. Thibodeau (Department of Speech Communication, University of Texas, Austin, TX 78712-1089)

The auditory enhancement effect was evaluated in normal and hearing-impaired listeners using a forward masking paradigm similar to that used by Viemeister and Bacon [J. Acoust. Soc. Am. **71**, 1502–1507 (1982)]. Masked thresholds for a 2000-Hz probe were compared between a standard condition in which the masker was a four-component harmonic complex including 2000 Hz, and an enhancing condition in which a three-component harmonic complex not including 2000 Hz preceded the four-component masker. The spectrum level of the masker was held constant at 85 dB SPL. For the normal listeners, the thresholds in the enhancing condition were approximately 30 dB greater than in the standard condition. For the hearing-impaired listeners, the enhancing condition resulted in only an 8- to 18-dB shift in threshold relative to the standard condition. One explanation for the enhancement effect in this case involves the adaptation of suppression in components surrounding the unadapted component such that there is actually an increase in intensity in the unadapted region. Suppression effects were also evaluated in both groups by measuring forward masked thresholds for a 2000-Hz probe as a function of a variable frequency suppressor tone that was added to a 2000-Hz masker. There was evidence of suppression for the normal listeners, particularly when the suppressor component was above the masker in frequency. However, for the hearing-impaired listeners, there was limited evidence of suppression as there was little change in masked threshold as a function of suppressor frequency.

Y21. Auditory temporal and reading disability. Betty U. Watson (Department of Speech and Hearing Sciences, Indiana University, Bloomington, IN 47405)

Deficits in auditory processing have been proposed by many authors as an etiology of some cases of specific reading disability [e.g., P. Tallal, Brain Lang. **9**, 182–198 (1980); E. B. Zurif and G. Carson, Neuropsychologia **8**, 351–361 (1970)]. Such deficits are hypothesized to impair precise perception of speech sounds, which in turn interferes with acquisition of sound-symbol correspondence when learning to read. This hypothesis was evaluated in six studies that used the Test of Basic Auditory Capabilities (TBAC) [Watson *et al.*, J. Acoust. Soc. Am. Suppl. 1 **71**, S73 (1982)] to assess temporal processing. The first study found significant differences in temporal processing between normal and learning-disabled high school students, but these differences were not replicated in a second study with a college sample. The next two studies found substantial relationships between measures of academic aptitude/intelligence and temporal processing. The last two studies report correlations between the TBAC and performance on a variety of phonic analysis tasks. Few significant relationships were found when the effects of intelligence or academic aptitude were controlled. Overall, this series of studies fails to support the causal role of temporal processing deficits in the etiology of specific reading disabilty [Work supported by NIH.]

Session Z. Physiological Acoustics V and Psychological Acoustics II: Animal Acoustics (Poster Session)

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Contributed Papers

All posters will be displayed from 8:00 a.m. to 12:00 noon. To allow contributors the opportunity to see other posters, contributors of all papers will be at their posters from 10:00 a.m. to 12:00 noon.

Z1. Melody recognition by an Atlantic bottlenose dolphin (*Tursiops truncatus*). James V. Ralston, Louis M. Herman, Humphrey N. S. Williams, John D. Gory, and Kristin K. Jerger (Kewalo Basin Marine Mammal Laboratory, University of Hawaii, 1129 Ala Moana Boulevard, Honolulu, HI 96814)

Previous studies with monkeys and songbirds indicate a limited capability to recognize interval or contour information in melodic sequences of discrete sinusoidal tones. In the present experiments, two different types of melodic sequences, either four-tone sequences with constant pitch or four-tone sequences with descending pitch, were projected underwater to a 12-year-old adult female Atlantic bottlenose dolphin. The initial training attempted to teach the dolphin to press specific paddles after presentation of stimuli. Instead, the dolphin began to whistle spontaneously after descending-pitch sequences and remained silent after constant-pitch sequences. Over the same trials, she responded at chance with paddle presses. The paddles were removed, and vocal responses were thereafter judged in real-time by "blind" listeners. In subsequent tests, the dolphin successfully transferred the vocal responses to several pitch-transposed stimuli drawn from within, as well as a full octave above, the pitch range of the original training stimuli. These results provide the first evidence of octave generalization in a nonhuman listener and suggest a robust form of interval or melody recognition in this dolphin species. Data will also be presented from a second experiment contrasting rising and falling pitch sequences. [Work suppored by ONR.]

Z2. Behavior specific vocalizations of two Atlantic bottlenose dolphins (*Tursiops truncatus*). Sarah Partan, Kristin Jerger, Mark Xitco, Herbert Roitblat, Louis Herman, Humphrey Williams, Michael Hoffhines, John Gory, and James Ralston (Kewalo Basin Marine Mammal Laboratory, University of Hawaii, 1129 Ala Moana Boulevard, Honolulu, HI 96814)

The vocalizations of two Atlantic bottlenose dolphins (*Tursiops truncatus*) were recorded during a study of behavioral mimicry. Underwater recordings were made on one channel of a stereo magnetic tape, while a behavioral trial-by-trial narrative was recorded onto a second channel. The tapes were analyzed by human listeners blind to the behavior performed. The vocalizations were divided into categories of pulsed versus nonpulsed (whistle) sounds. There was a significant dependency between vocalizations produced and behaviors performed ($\chi^2 = 320.48$, $df = 60$, $p < 0.00001$). This finding suggests that the dolphins may communicate behavior specific information in their vocalizations. The results of current spectrographic and further behavioral studies will also be presented, with the objective of determining whether the vocalizations are communicative or are artifacts of the behavior.

Z3. Delayed matching-to-sample by an echo-locating bottlenose dolphin. H. L. Roitblat, Ralph H. Penner, and Paul E. Nachtigall (Department of Psychology, The University of Hawaii at Manoa, 2430 Campus Road, Honolulu, HI 96822 and The Naval Ocean Systems Center, Kailua, HI 96734)

An Atlantic bottlenose dolphin (*Tursiops truncatus*) was trained to perform three-alternative delayed matching-to-sample while wearing eye cups to occlude its vision. Samples and comparison stimuli consisted of small aspect-independent three-dimensional stimuli. Observing responses, in the form of echo-location clicks, were recorded during the presentation of the sample as well as during the presentation of the comparison stimuli. The location, duration, and rate of clicks to the sample and to each comparison stimulus was recorded. Over 48 sessions of testing, choice accuracy averaged 94.4% correct. Stimuli varied in the number of clicks necessary to identify it as the sample. Examination of the number of echolocating signals emitted and their distribution over locations reveals a complex decision making process in which stereotyped sequences are combined with contingent item identification to yield accurate performance. A model for the dolphin's decision-making processes is described.

Z4. Voco-auditory functions of the chimpanzee. Shozo Kojima (Primate Research Institute, Kyoto University, Inuyama, 484 Japan), Itaru F. Tatsumi (Tokyo Metropolitan Institute Gerontology, 35-2 Sakae-cho, Itabashi-ku, Tokyo, 173 Japan), Shigeru Kiritani, and Hajime Hirose (RILP, University of Tokyo, 7-3-1 Hongo, Bunkyo-ku, Tokyo, 113 Japan)

Auditory sensitivity, frequency, and intensity difference thresholds were measured in chimpanzees. Chimpanzees had a W-shaped auditory sensitivity function. That is, they were less sensitive to the midfrequency range of 2–4 kHz than to 1–8 kHz (the best frequencies). The frequency and intensity difference thresholds at 1 kHz and 70 dB SPL were about 10 Hz and about 1.5 dB, respectively. In the perception of vowels, reaction time for vowel discrimination was longer for [i]–[u] and [e]–[o] pairs than for other vowel pairs. This vowel perception can be explained by the W-shaped auditory sensitivity function. In the perception of stop consonants, distinction of the place feature was more difficult than that of the voicing feature. Preliminary experiments indicated that chimpanzees show vocal tract normalization effects for vowels and phoneme boundary effects for stop consonants.

Z5. Macaques show context effects in speech perception. Erica B. Stevens, Patricia K. Kuhl, and Denise M. Padden (Department of Speech and Hearing Sciences, University of Washington, Seattle, WA 98195)

The phonetic distinction between the stop consonant /b/ and semivowel /w/ is cued primarily by duration of the initial formant transition. Syllable duration, however, is a context that influences human adults' and infants' perception of this distinction. For example, when overall syllable duration is long (296 ms), infants show poor discrimination of syllables with transition durations of 16 and 40 ms, but good discrimination of syllables with transition durations of 40 and 64 ms. However, when overall syllable duration is short (80 ms), the opposite is true [P. D. Eimas and J. L. Miller, Science **209**, 1140–1141 (1980)]. The current study addresses whether rhesus and Japanese macaques' perception of the /b-w/ distinction is also influenced by syllable duration. Six CV syllables used in the Eimas and Miller (1980) study were presented to five macaques using a same–different discrimination paradigm. Results showed that macaques' pattern of discrimination was similar to the pattern shown by infants, thus supporting an auditory-based mechanism for the effect of syllable duration in phonetic perception. [Work supported by NSF and NIH.]

Z6. Tone-on-tone masking in the goldfish: The effects of signal phase for signal and masker at 500 Hz. R. R. Fay (Parmly Hearing Institute, Loyola University of Chicago, 6525 North Sheridan Road, Chicago, IL 60626)

Thresholds for detecting 20-ms, 500-Hz tone bursts (7-ms rise/fall times) in the presence of a 500-Hz masker were measured in goldfish using classical respiratory conditioning with a tracking psychophysical procedure. Thresholds were obtained using a continuous masker, and with the signal temporally centered in a 200-ms gated masker. Thresholds were determined as a function of the phase angle at which the signal was added to the masker. For the continuous masker, the just-detectable sound-pressure increment (at 0-deg phase angle) was 0.15 dB, and the just-detectable decrement (at 180 deg) was -0.22 dB. For the gated masker, the thresholds were 1.75 and -2.4 dB, for the in-phase and phase-inverted signals, respectively. Varying signal phase between 0 and 90 deg resulted in a progressively declining increment threshold, reaching about 0.5 dB at 90 deg. Since the threshold increment is not constant as a function of signal phase, some other cue(s) must underlie detection. Spectral "splatter" is not likely the cue since the increment threshold varied little when signal rise time was reduced to zero, or when the signal was bandpass filtered (96 dB/oct) at 500 Hz. This suggests that goldfish directly detect a transient phase shift occurring in the masker waveform when the out-of-phase signal is added. [Work supported by NINCDS Program Project Grant.]

Z7. The response of a hot film anemometry system and the mechanosensory lateral line of a teleost fish to the local flow field of a dipolar source. Sheryl Coombs and John Janssen (Parmly Hearing Institute and Department of Biology, Loyola University of Chicago, IL 60626)

A hot film anemometry system (TSI, inc.) was evaluated for its potential use in characterizing lateral line stimuli by measuring its response to a sinusoidally vibrating sphere over a wide range of frequencies, intensities, and distances from the sphere. The amplitude response of the anemometer probe to signals along the axis of vibration was found to be linear over a 50-dB range for frequencies from 10–200 Hz and responses to displacement levels as low as 10^{-9} m were recorded. The measured attenuation of the signal with distance was also linear over this frequency range out to distances of six times the radius ($=3$ mm) of the sphere, and followed the predicted falloff rate, $1/r^3$ (where $r=$ distance from sphere center), for a dipolar source [Harris and van Bergeijk, J. Acoust. Soc. Am. **34**, 1831 (1962)]. Behavioral measures of lateral line sensitivity in a benthic fish, the mottled sculpin (*Cottus bairdi*), also attenuated at the rate of $1/r^3$ for a wide range of stimulus frequencies. This attenuation rate can be predicted for benthic fish which, because they are coupled to the substrate, do not move in the flow field, but not for midwater fish that are neutrally buoyant and coupled to the surrounding water. [Work supported by NIH].

Z8. Midline and lateral field sound localization by the grasshopper mouse, *Onychomus leucogaster*. Jack B. Kelly and Caren L. Furlonger (Laboratory of Sensory Neuroscience, Psychology Department, Carleton University, Ottawa, Ontario K1S 5B6, Canada)

Comparative studies of sound localization have placed almost exclusive emphasis on obtaining minimum audible angles for sources positioned symmetrically around midline. Little attention has been paid to the ability of animals to localize sounds within the lateral fields and reports of minimum audible angles for off-midline positions are rare. A recent study by Kavanagh and Kelly [Behav. Neurosci. **100**, 200–205 (1986)] indicates that the albino rat has great difficulty localizing a brief acoustic stimulus in the lateral fields, a finding that is unexpected from estimates of midline acuity. The present study was undertaken to determine whether other small mammals might experience similar difficulty. Minimum audible angles were obtained for 0, $+60$, and -60 deg azimuth using a broad-spectrum single click in a two-choice spatial task. The minimum audible angles as defined by 75% performance levels were 28 deg for midline and 48.5 deg for the lateral fields indicating that the grasshopper mouse does have difficulty with lateral field localization but not to the same extent as the albino rat. [Work supported by NSERC.]

Z9. Learned perceptual categories for complex vocal signals in budgerigars (*Melopsittacus undulatus*). Robert J. Dooling, Susan D. Brown, Thomas J. Park, and Kazuo Okanoya (Department of Psychology, University of Maryland, College Park, MD 20742)

Operant conditioning and multidimensional scaling (MDS) procedures were used to study auditory perception of complex vocal signals in budgerigars. Response latencies were anlyzed by MDS to produce a spatial arrangement of these complex sounds reflecting the perceptual organization. Both normal and isolate-reared budgerigars and humans show perceptual categories which correspond closely to the major functional and acoustic categories of species-specific calls. Birds sharing the same learned call show perceptual categories for the calls of individual birds. Humans and other budgerigars unfamiliar with these same learned calls fail to group calls by individual. These results suggest that both general and special auditory mechanisms are involved in the formation of perceptual categories for species-specific vocal signals in budgerigars. Additional tests with human speech sounds suggest that the general auditory processing mechanism in budgerigars are quite sophisticated. [Work supported by NIH.]

Z10. Mysticete whale sounds and human speech. Kevin Chu and R. Stuart Mackay (Biology Department, Boston University, Boston, MA 02215 and San Francisco State University, San Francisco, CA 94132)

Toothed whales produce complex sounds, in some cases with two or more simultaneous sources in the head [R. S. Mackay and H. Liaw, Science **212**, 676–678 (1981)] but mysticete vocalizations are said to be simple. Frequency analysis of recordings of humpback whales *Megaptera novaeangliae* on Silver Bank, Dominican Republic, produced sound spectrograms where certain pairs of components could both be present but with one increasing in frequency while the other decreased, or one alone could appear, or both could change in the same direction. This action is familiar in the formants of human speech, and some similarity of modulation mechanism is possible (though whales have no vocal cords). It does not seem possible that these recordings involved two animals singing in unison because timing remained the same for wide shifts of receiving point, which changed the relative distance to members of any source pair.

Session AA. Physiological Acoustics VI and Psychological Acoustics III: Localization and Binaural Hearing (Poster Session)

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Contributed Papers

All posters will be displayed from 8:00 a.m. to 12:00 noon. To allow contributors an opportunity to see other posters, contributors of papers AA1 through AA7 will be at their posters from 8:30 to 10:00 a.m. contributors of papers AA8 through AA13 will be at their posters from 10:00 to 11:30 a.m.

AA1. The Franssen effect and the localization plausibility hypothesis. Brad Rakerd (Department of Audiology and Speech Sciences, Michigan State University, East Lansing, MI 48824) and William Morris Hartmann (Department of Physics, Michigan State University, East Lansing, MI 48824)

The Franssen effect is obtained with two loudspeakers in a room. If a sine tone is abruptly turned on at the left loudspeaker, then slowly faded off while the right loudspeaker is slowly faded on, a listener will judge that the tone continues to come from the left loudspeaker, even though the left loudspeaker is not sounding at all. Our own studies of localization of sound in rooms have led to a principle of localization called the "plausibility hypothesis." One of the predictions of this hypothesis is that in an anechoic room the Franssen effect should fail [B. Rakerd and W. M. Hartmann, J. Acoust. Soc. Am. **78**, 524–533 (1985)]. Experimental studies are reported using the Franssen stimulus both in an ordinary room and in an anechoic room. The results of the experiment support the prediction. [Work supported by the National Institutes of Health.]

AA2. Acoustic origins of individual differences in sound localization behavior. Elizabeth Wenzel, Frederic Wightman, Doris Kistler, and Scott Foster (NASA-Ames Research Center, Moffett Field, CA 94035, Department of Psychology and Waisman Center, University of Wisconsin, Madison, WI 53705, and Crystal River Engineering, 12350 Wards Ferry Road, Groveland, CA 95321)

Human listeners vary widely in their ability to localize unfamiliar sounds in an environment devoid of visual cues. Our research, in which blindfolded listeners give numerical estimates of apparent source azimuth and elevation, suggests that individual differences are greatest in judgments of source elevation; listeners are uniformly accurate when judging source azimuth. The pattern of individual differences is the same for free-field sources and for simulated free-field sources presented over headphones. Simulated free-field sources are produced by digital filtering techniques which incorporate the listener-specific, direction-dependent acoustic effects of the outer ears. Two features of this data bear on the question of the origin of individual differences in elevation accuracy: (1) a listener's accuracy in judging source elevation can be predicted from an analysis of the acoustic characteristics of the listener's outer ears; (2) the pattern of elevation errors made by one listener (A) can be transferred to another listener (B) by presenting to listener B the simulated free-field sources derived from the outer-ear acoustics of listener A. Thus it is believed that many of the individual differences in localization behavior are traceable to individual differences in outer-ear acoustics. The data have important implications for the study of localization in both basic and applied contexts. [Work supported by NASA, NSF, and USAF-AAMRL-AFSC.]

AA3. Prefiltering method for a head-related stereophonic system. Takayuki Mizuuchi, Kaoru Okabe, Hareo Hamada, and Tanetoshi Miura (Tokyo Denki University, 2-2 Kanda-Nishiki-cho, Chiyoda-ku, Tokyo, 101 Japan)

The prefiltering scheme of the usual head-related stereophonic system using two loudspeakers for reproduction [M. R. Schroeder *et al.*, J. Acoust. Soc. Am. **56**, 1195 (1974)] was modified. In addition to the usual filtering scheme, an additional filtering stage is introduced that converted the frontal incident characteristics of the dummy head into those for individual listeners. Using this filtering scheme, the exact reproduction of the frontal sound image becomes possible, which is difficult when the sound recorded through a dummy head is played back. The ability of this modified system to reproduce the original sound localization by a listening test in an anechoic chamber is evaluated. In the test, both the conventional scheme and the new one were evaluated. As a result, the localization of the frontal sound image was reproduced exactly using the new system, and the conventional system failed in this reproduction. It was also confirmed that sound from other directions in the three-dimensional space was almost perfectly reproduced using the new system.

AA4. Echo suppression or localization masking? Pierre L. Divenyi (Speech and Hearing Research, V. A. Medical Center, Martinez, CA 94553)

A brief dichotic conditioner (C) has been shown to effectively disrupt lateralization of a brief probe (P) presented after a short interval (4–7 ms, onset to onset) with an interaural time delay that is perfectly audible in the absence of the C [P. M. Zurek, J. Acoust. Soc. Am. **66**, 1750–1757 (1980)], even when the C and the P are different sounds [P. L. Divenyi and J. Blauert, in *Auditory Processing of Complex Sounds*, edited by W. A. Yost and C. S. Watson (Erlbaum, Hillsdale, NJ, 1987), pp. 146–155]. An echo suppression mechanism responsible for this effect would predict (1) suppression to be strongest when the C and the P are identical and to decrease monotonically as the spectra of the two sounds are made different, and (2) a monotonic falloff of suppression when the temporal separation between the two sounds is increased beyond a certain minimum. In the present experiments, the effects of frequency separation and temporal separation between a narrow-band P centered at 2 kHz and a C were systematically explored. Contrary to the predictions, low-frequency ($0.8 < 1.3$ kHz) C's were more effective in suppressing the lateralization of the P than those closer to the P frequency, and lateralization performance was nonmonotonically related to temporal separation between C and P. The results suggest that a dichotic stimulus with a relatively high *localization strength* could "mask" the localization of another, subsequent dichotic stimulus. [Work supported by the Veterans Administration.]

AA5. Dependence of interaural time relationships in the cat upon sound source direction and frequency. J. E. Hind, A. D. Musicant, J. C. K. Chan, and R. K. Kochhar (Department of Neurophysiology, University of Wisconsin Medical School, Madison, WI 53706)

Using a probe microphone surgically implanted near each eardrum of an anesthetized cat, the time waveform was recorded in both ears in response to a click as the azimuth and elevation of a loudspeaker were varied. Two forms of interaural time delay (ITD) were calculated. Arrival time (AT) for each ear was based upon the leading edge of the click waveform and the first measure of ITD was taken as the difference in AT for the two ears. The FFT analysis of the time waveforms also yielded the interaural phase difference (IPD) for the individual frequency components. The phase-derived ITD was calculated as IPD/ω and enabled study of the dependence of ITD upon frequency. Both measures of ITD varied systematically with azimuth and elevation and reached values greater than would be expected from the interaural distance. Kuhn [*Directional Hearing*, edited by W. Yost and G. Gourevitch, (Springer, Berlin, 1987), Chap. 1] has shown for the human that ITDs for low and high frequencies are in the ratio of 3:2, respectively. The present study yields similar findings in the cat and confirms and extends earlier work.

AA6. Lateralization predictions for high-frequency binaural stimuli. Richard M. Stern, Glenn D. Shear, and Torsten Zeppenfeld (Department of Electrical and Computer Engineering, Carnegie Mellon University, Pittsburgh, PA 15213)

The position-variable model [R. M. Stern, Jr. and H. S. Colburn, J. Acoust. Soc. Am. **64**, 127–140 (1978); G. D. Shear and R. M. Stern, J. Acoust. Soc. Am. Suppl. 1 **81**, S27 (1987)] is extended to describe the subjective lateral position of amplitude-modulated tones and bandpass noise, as well as other complex stimuli that are presented within spectral regions at which the binaural system appears to be unable to make use of cycle-by-cycle interaural temporal differences. Predictions of the model are based on the centroid of the cross correlation of the hypothetical auditory-nerve response to the stimuli, which is either calculated using analytical techniques or simulated numerically. The model of auditory-nerve activity, which is typically used to describe the response to stimuli at lower frequencies, also extracts envelopes of higher frequency stimuli, as discussed previously by Colburn and Esquissaud. This information appears to be useful in predicting the lateral position of such high-frequency stimuli. Preliminary results indicate that the model is able to describe most of the ways in which the laterality of high-frequency binaural stimuli with low-frequency envelopes depends on modulation frequency, carrier frequency, and other stimulus parameters. The model also predicts the relative salience of interaural temporal cues at different frequencies. [Work supported by NSF.]

AA7. Detection and lateralization based on interaural temporal disparities: Time or phase? Leslie R. Bernstein and Constantine Trahiotis (Center for Neurological Sciences and Department of Surgery, University of Connecticut Health Center, Farmington, CT 06032)

This presentation will address whether the binaural auditory system's sensitivity to interaural temporal disparities is more appropriately characterized as a sensitivity to physical differences, across the ears, of *time* or *phase* per se. In two recent studies [W. A. Yost and R. H. Dye, J. Acoust. Soc. Am. **83**, 1846–1851 (1988); W. A. Yost, J. Acoust. Soc. Am. **70**, 397–409 (1981)], Yost and Dye measured thresholds for detection of, and extent of, laterality produced by interaural temporal disparities of pure tones between 200 and 5000 Hz. Because performance in both tasks appeared to be constant across frequency for a given interaural *phase* difference rather than for a given interaural *time* difference, Yost and Dye argued that phase is the most appropriate independent variable. This notion is inconsistent with several other sets of data obtained with complex signals and with pure tones. Furthermore, the constraints that the processing of interaural phase per se imposes on modern cross-correlation-based models of binaural hearing stand in contrast to the more parsimonious explanations based on interaural time.

AA8. Transient latency in masking level difference. Hisashi Kado, Shigeru Chiba, Kohzo Ohta, Hajime Miura (Electrotechnical Laboratory, 1-1-4 Umezono, Tukuba, 305 Japan), and Masaaki Fukumoto (University of Electro-Communications, 1-5-1 Chofugaoka, Chofu, 182 Japan)

The threshold was measured for a signal S_0 of 100 ms long at various timing points in a noise burst by a sequential searching procedure. The noise burst lasted from $t = -1000$ ms to $t = 1000$ ms. At $t = 0$ ms, the noise was changed from N_0 to N_π. The threshold was expected to decrease just after $t = 0$ ms. The latency was defined as a time constant of the decreasing characteristic of the threshold. In the case of S_0 being a pure tone, the results showed artifacts at the N_0 area and it was hard to define the latency. In the case of S_0 being a vowel (female /a/, /i/ and male /i/), the latency was about 100 ms when subjects were required to identify the signal, and the latency was about 10 ms when subjects were required to detect the signal. The equalization and cancellation model could not explain the 10-ms latency, because it is not reasonable to assume such fast feedback in the model. It is speculated that both the addition and subtraction of dichotic and/or diotic signals are sent to a higher level of the hearing system in such fast latency.

AA9. Discrimination of the orientation of a natural sound source. Pamela Zerne, David Perrott, John Cote, and Charles Lira (Psychoacoustics Laboratory, California State University, Los Angeles, CA 90032)

The ability to discriminate the direction that a speaking individual is facing (i.e., toward, away, to the right, or to the left relative to the listener) was explored in a free-field environment under monaural and binaural listening conditions with relative distances ranging between 1.5–12.0 m. All sessions were conducted under double-blind conditions. Performance under binaural conditions clearly exceeded chance at all orientations and distances. The ability to resolve that the speaker was facing right or left was significantly impaired when binaural cues were eliminated. This surprising ability to resolve the relative orientation of a human speaker in the right–left dimension cannot readily be explained by current theories of auditory spatial processes. [Work supported by NSF and NIH.]

AA10. Utilization of spatially correlated auditory information in localization of visual targets. Kourosh Saberi, David R. Perrott, and Sandra M. Pacheco (Psychoacoustics Laboratory, California State University, Los Angeles, CA 90032)

Facilitation of a visual search task through presentation of spatially correlated and uncorrelated binaural and monaural cues was determined for five subjects using a two-alternative, forced-choice paradigm. Trains of 10-ms clicks were spatially correlated with a visual target (0.496° visual angle) randomly located in a $\pm 120°$ azimuth $\pm 46°$ vertical spherical visual field and presented to subjects in two conditions (0°, $\pm 46°$ vertical). Visual search time was also determined for monaural conditions in the $\pm 46°$ vertical $\pm 120°$ azimuth paradigm. Results indicate a dramatic reduction in search time for spatially correlated cues from approximately 15% at 0° to 40% at $\pm 120°$ azimuth in the 0° elevated condition, and from 35% at 0° to 50% at $\pm 120°$ azimuth in the $\pm 46°$ elevated condition. The search time function for the monaural condition is surprisingly more similar to that of the spatially correlated binaural one, particularly in the center regions. Data indicate that loss of search time at the periphery in the uncorrelated condition is mostly due to the search strategy employed and not head movement delays. Practical implications will be discussed. [Work supported by NSF and NIH.]

AA11. Minimum audible movement angle thresholds for broadband noise: Effects of the initial location of the source. David R. Perrott, Carol L. Manligas, and Sandra Pacheco (Psychoacoustics Laboratory, California State University, Los Angeles CA 90032)

Minimum audible movement angle (MAMA) thresholds were determined for a source producing a broadband noise (500–8000 Hz) moving at a constant angular velocity (20°/s). Nine regions of the field were sampled: 0°, 10°, 20°, 40°, and 80° to the subject's right and left. In agreement with earlier research that employed either static sources or "simulated motion," acuity was best when the source was at 0° azimuth (1.1°) and poorest at the most lateral position (3.1°–3.5°). The ability to detect motion of a source emitting a broadband noise is clearly superior to performance obtained with tonal signals whether the latter paradigms include "actual" or "simulated" motion. In the more extreme lateral regions of the field, performance was actually better than that previously reported under static localization conditions. Vertical azimuths, ranging from 0°–87.5°, were also examined. Detection of motion was relatively insensitive to this latter parameter. [Work supported by NSF and NIH.]

AA12. Rotating tones and minimum audible movement angle. David R. Perrott and B. Meier (Psychoacoustics Laboratory, California State University, Los Angeles, CA 90032)

Minimum audible movement angle (MAMA) thresholds were determined for four subjects using a two-alternative, forced-choice adaptive paradigm. The motion was *simulated* by presenting a 200 to 1500-Hz tone (f_0) to one ear and a second tone (f_1) of a different frequency to the other ear. With interaural frequency differences (IFD) of 0.5 Hz, and f_0 at or below 600 Hz, MAMA thresholds ranged between 10°–15°. Additional tests conducted in this frequency region indicate that the continuous shifts in phase generated by small IFDs produce performance in excellent agreement with those obtained in free-field localization tasks in which *real* and *simulated* motion has been considered. However, thresholds increase very rapidly at higher frequencies (e.g., MAMA exceeds 80° with an f_0 of 850 Hz). There appears to be little agreement between the current "dichotic" rotating tone paradigm and other motion discrimination functions at these higher frequencies. The implications of these results will be discussed. [Work supported by NSF and NIH.]

AA13. Click lateralization by multiple sclerosis patients. Miriam Furst (Department of Electronic Systems, Faculty of Engineering, Sackler School of Medicine, Tel Aviv University, Tel Aviv 69978, Israel), Sara Eyal, and Amos D. Korczyn (Department of Physiology and Pharmacology, Sackler School of Medicine, Tel Aviv University, Tel Aviv 69978, Israel)

Lateralization of dichotic clicks can be tested with gradually increasing interaural level differences (ILD) and interaural time differences (ITD). However, the lateralization of dichotic clicks with ITD is limited to small ITDS only (< 1 ms). The ability to lateralize dichotic clicks was tested in multiple sclerosis (MS) patients with normal audiograms. Two kinds of psychoacoustical experiments were employed: (1) a matching experiment in which the subject was asked to match the perceived positions of two click trains, where one included dichotic clicks wlth ILD and the other included dichotic clicks with ITD; (2) a positional jnd experiment in which the subject was asked to determine the difference in position of two successive click trains. Two reference positions were tested, the head center (ITD = 0 and ILD = 0) and midway between the center and the ear (ITD = 0.8 ms and ILD = 0 or ILD = 20 dB and ITD = 0). For each reference the experiment was performed first with control on ITD and then with control on ILD. From a group of 15 MS patients, seven performed poorly in the positional jnd experiment when the control was on ITD, and normally when the control was on ILD. Those subjects also reported that the position of each dichotic click with ITD < 1 ms was perceived at the head center, and, therefore, they had difficulties in performing the matching experiment. Brainstem auditory potentials (BAP) evoked by dichotic clicks with different ILDs and ITDs were measured in all the above MS patients. Principal component analysis of the measured waveforms indicates an abnormal behavior as a function of ITD in patients who performed abnormally in the psychoacoustical experiments. The anaylsis of the waveforms of all patients as a function of ILD resembles the normal behavior.

WEDNESDAY MORNING, 16 NOVEMBER 1988

KAUAI ROOM (EAST END), 8:00 TO 10:00 A.M.

Session BB. Speech Communication V: Production, Part A (Poster Session)

Kenneth N. Stevens, Cochairman
Research Laboratory of Electronics
Massachusetts Inst. of Technology
Cambridge, Massachusetts 02139

Hisayoshi Suzuki, Cochairman
Faculty of Engineering
Shizuoka University
3-5-1 Jouhoku
Hamamatsu, 432 Japan

Contributed Papers

Posters should be set up before 8:00 a.m. All posters will be displayed from 8:00 to 10:00 a.m. To allow contributors the opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:00 to 9:00 a.m. and contributors of even-numbered papers will be at their posters from 9:00 to 10:00 a.m.

BB1. Measures of laryngeal adduction. Ronald C. Scherer and Vernon J. Vail (Recording and Research Center, The Denver Center for the Performing Arts, Denver, CO 80204)

The degree of closeness of the arytenoid cartilages (laryngeal adduction) plays an important role in vocal function and quality. In this paper, adduction for steady vowels was varied between voiced qualities of breathy (greater abduction) to normal to constricted (greater adduction). The superior view of the larynx of various subjects was videotaped and dimensions of laryngeal adduction were measured. The measurements were compared with values of the abduction quotient [I. R. Titze, J. Acoust. Soc. Am. 75, 570–580 (1984)] obtained from simultaneous recordings of the electroglottograph signal and with a measure of electroglottograph waveform width. The width measure strongly correlates with the abduction quotient. Results suggest that the adduction measures may be useful for clinical and research purposes. [Work supported by NIH.]

BB2. Simultaneous modeling of EGG, PGG, and glottal flow. B. Cranen (Department of Phonetics, Nijmegen University, P. O. Box 9103, 6500 HD Nijmegen, The Netherlands)

For obtaining insight into basic aspects of voice quality, an adequate parametrization of the voice source is mandatory. Theoretical studies have shown that changes of the glottal geometry in the direction of flow are an essential prerequisite for self-oscillation of the vocal folds. Thus it is attractive to use stylized glottal shapes as a basis for voice source parametrization, and try to extract the geometry parameters (semi-) automatically from indirect measurements like the electroglottogram (EGG) and photoglottogram (PGG). Titze [J. Acoust. Soc. Am. **75**, 570–580 (1984)] presented a model with which he simulated simultaneous measurements of PGG and EGG. It is thought that EGG and PGG are not sufficient for inferring all relevant aspects of glottal geometry. In this paper, it is shown that glottal flow supplies additional information that can be used to impose extra restrictions on the model parameters. Simulations show that when simultaneously modeling EGG, PGG, and glottal flow, it is crucial to assume that the opening of the vocal folds occurs more gently than the closing. Implications of this finding for the description of vocal fold movements in modeling EGG and PGG are discussed.

BB3. Various laryngeal mechanisms in controlling the voice fundamental frequency. Kiyoshi Honda (Department of Electronics, Kanazawa Institute of Technology, Kanazawa-Minami, 921 Japan)

It is generally believed that the muscular control for the voice fundamental frequency ($F0$) is performed mainly by the cricothyroid. However, it has been noted in several studies that the whole $F0$ range produced by humans cannot be obtained by cricothyroid activity alone. This study reports variations in the EMG activity of some laryngeal muscles associated with $F0$ variations in word utterances and the speculated mechanisms of $F0$ control by these muscles. First, the cricopharyngeus (CP) exhibits an inverse correlation with $F0$, suggesting that this muscle can rotate the cricoid cartilage inversely. Second, the lateral cricoarytenoid (LCA) shows the highest correlation with $F0$, indicating its function in adjusting the effective length of the vocal fold vibration. Third, the oblique part of the cricothyroid (CT) has a higher correlation with $F0$ than the rectal part, and it is speculated that the rectal part produces the fast $F0$ change associated with laryngeal articulation and that the oblique part functions to expand the $F0$ range independently of articulatory gestures. These speculated mechanisms explain the observed nonlinear relation between vocal fold length and $F0$. [A part of this work was done at Haskins Laboratories, New Haven, CT.]

BB4. The fundamental frequency–subglottal pressure ratio in speech. Louis Boves and Helmer Strik (Department of Phonetics, Nijmegen University, P. O. Box 9103, 6500 HD Nijmegen, The Netherlands)

It is known that subglottal pressure P_s is a major factor in the control of fundamental frequency ($F0$) in speech. Yet, the details of this relation remain unclear. Estimates of the $dF0/dP_s$ ratio obtained from speech and special phonation tasks yield values between 5 and 13 Hz/cm aq. "Push-in-the-stomach" experiments, on the other hand, tend toward values of 3–5 Hz/cm aq. The experiments discussed here aim at resolving this discrepancy. Simultaneous registrations have been made of subglottal and oral pressure, laryngeal EMG, EGG, and lung volume during a number of phonation tasks. Present results suggest that in the push-in-the-stomach experiments $F0$ lags P_s to such an extent that the full effect of changes in P_s is not attained before the subject starts compensating (in one way or another) for the disturbance in P_s. Thus these experiments tend to underestimate the $dF0/dP_s$ ratio in undisturbed phonation tasks. Implications for the explanation of declination in speech will be discussed. [Research supported by the Foundation for Linguistics, funded by N. W. O.]

BB5. A control mechanism of voice intensity in relation to expiratory and laryngeal functions. Reiko M. McCafferty (6020 Kathryn S.E., No. 30, Albuquerque, NM 87108), Yuki Kakita, and Morihiko Nakata (Department of Electronics, Kanazawa Institute of Technology (KIT), Kanazawa-Minami, 921 Japan)

A model explanation of controlling voice intensity in relation to expiratory and laryngeal functions is reported. Using the phonatory function analyzer PS-77 (Nagashima Medical Instruments, Tokyo), a log-linear relationship between the vocal intensity I^* (dB) and the mean airflow rate $U(\mathrm{cm}^3/\mathrm{s})$ was obtained during a stepwise increment of intensity with a constant pitch. More precisely, a piecewise linear relationship with an S-shaped pattern was obtained in an I^* vs U^* display, where $U^* = 20 \log(U/U_0)$. At most, three line segments with different slopes are necessary to approximate these relationships. To examine, theoretically, the controlling mechanism of the intensity for the different slope values in I^* vs U^* display, a simple intermittent triangular wave was assumed as the glottal sound. Using the model waveform, the vocal intensity can be expressed by a functional relationship with the mean airflow rate U and the open quotient. Based on this functional relationship and speculation on the physiology voice production, the contribution of laryngeal control and expiratory control on vocal intensity is discussed. Results of the experiment indicate that laryngeal control is more dominant in the middle intensity region than in either of the higher or the lower intensity regions. These findings may contribute to developing an effective means of noninvasive voice assessment.

BB6. Control of fundamental frequency and intensity in running speech. Helmer Strik and Louis Boves (Institute of Phonetics, Nijmegen University, P. O. Box 9103, 6500 HD Nijmegen, The Netherlands)

This study explores the control of fundamental frequency ($F0$) and intensity level (IL) in running speech. To that end, simultaneous recordings of speech, electroglottogram, lung volume, sub- and supraglottal pressure (P_{sb} and P_{sp}), and EMG activity of cricothyroid, vocalis, and sternohyoid were obtained for three Dutch subjects. After preprocessing, the data were subjected to a stepwise multiple regression analysis. Here, $F0$ and IL were used as dependent variables. Results show that the pressure signals are the most important factors in the control of both $F0$ and IL. However, results are dependent on the method of analysis used. On sentence level, P_{sb} appears to be the most important mechanism controlling $F0$, while on word level P_{sp} and muscle activity become more important. For IL, on sentence level P_{sb} and P_{sp} seem to contribute equally to its control via transglottal pressure. But on the level of individual words P_{sp} alone explains approximately 80% of the total variance of IL. [Research supported by the Foundation of Linguistics, funded by N. W. O.]

BB7. Gestural aggregation in speech. Kevin Munhall (Department of Communicative Disorders, Elborn College, University of Western Ontario, London, Ontario N6G 1H1, Canada) and Anders Löfqvist (Department of Logopedics and Phoniatrics, Lund University, Sweden and Haskins Laboratories, New Haven, CT 06511)

It is well known that the units of speech are not produced strictly sequentially but overlap with each other; this phenomenon has been referred to as coarticulation, coproduction, or blending. Thus the vocal tract shape at any one time during speech represents an aggregate of gestures associated with different production units. The present experiment examines one intra-articulator example of temporal overlap, namely, the combination of successive opening and closing gestures in the larynx. Two subjects produced the utterance "Kiss Ted" at several different speaking rates and with stress on the first or second word. Laryngeal abduction–adduction was monitored using transillumination and fiberoptic video recording. At slow rates, two separate opening gestures occur associated with the fricative and the stop, respectively. At fast rates, a single smooth gesture is seen. At intermediate rates, partially overlapping gestures are found. These findings suggest that the two underlying gestures are being blended into a single aggregate and that the aggregation operation may be some type of summation. It is hypothesized that blending of overlapping gestures is a general phenomenon in speech production that accounts for commonly observed types of coarticulation. [Work supported by NINCDS Grants NS-13617 and NS-13870 to Haskins Laboratories.]

BB8. Acoustic description of rehearsal effects on the voice. Dennis Ingrisano (Speech and Hearing Research Laboratory, University of Northern Colorado, Greeley, CO 80639) and Cynthia Chicouris (Voice Laboratory, University of Redlands, Redlands, CA 92691)

The vocal characteristics of six amateur singers were recorded longitudinally as the performers prepared for a Carnegie debut. The demands of rehearsals increased as the singers neared the performance date. Recordings were taken before and after rehearsal for a 3-month time period. Voice samples derived from reading material and sustained phonation were analyzed for frequency and amplitude perturbation; noise components associated with vocal use were also estimated. Results are discussed with reference to vocal fold overuse in this subject group and in previously reported literature [N. Punt, J. Laryngol. **97**, 13–17 (1983); P. Ward and G. Berci, Laryngoscope **92**, 13-77-1382 (1982)].

BB9. Physiologic factors in vocal vibrato. Thomas Shipp, E. Thomas Doherty (Speech Research Laboratory, VA Medical Center, San Francisco, CA 94121), and Stig Haglund (Department of Otolaryngology, Karolinska Hospital, Stockholm, Sweden)

Vocal vibrato when produced on a sustained vowel with constant subglottal air pressure is made up principally of fundamental frequency oscillations about the target pitch. This type of phonation was studied in two professional and one amateur singer, while EMG activity was sampled from their cricothyroid muscle (CT) unilaterally. Singers initiated phonation at a designated frequency without vibrato and then changed to their customary vibrato production. Coincident with the onset of vibrato, the EMG activity from the CT changed from a relatively stable interference pattern to one of quasiperiodic bursts and attenuation. Bursts immediately preceded initiation of upward $F0$ for the vibrato wave while attenuation preceded downward $F0$ movement. Further, the average integrated EMG value over one vibrato cycle equaled the EMG level for straight tone production at the same frequency. These data provide evidence for a metering system of neuromuscular discharges that equate CT activity and $F0$ mean whether during production of straight tone or vibrato oscillations.

BB10. A model for neurologic sources of vocal instability. Ingo R. Titze (Voice Acoustics and Biomechanics Laboratory, Department of Speech Pathology and Audiology, University of Iowa, Iowa City, IA 52240 and The Recording and Research Center, Denver Center for the Performing Arts, Denver, CO 80204)

The activation and relaxation characteristis of thyroarytenoid and cricothyroid muscle twitches are used as basic building blocks to simulate a tetanic contraction in the laryngeal muscles. Approximately 100 motor units are used with random phase relationships and randomized firing rates. The fundamental frequency of the voice is calculated from the simulated tensions, and it is shown that jitter and shimmer vary inversely with mean motor unit firing rate and with the number of motor units within a given muscle. Jitter and shimmer vary directly with the coefficient of variation of motor unit size. Some low-frequency variations in F_0 (tremor-like variation) are correlated with the mean firing rate of large single motor units. [Work supported by NINCDS Grant No. NS24409-02.]

BB11. Prosodic changes in speech following brain damage: Acoustic and neuroradiographic measures. D. Van Lancker, W. Hanson, C. Jackson, A. Lanto, E. J. Metter, and J. Cummings (Audiology and Speech Pathology, Sepulveda VA Medical Center, 16111 Plummer Street, Sepulveda, CA 91343)

Two prosodic parameters, fundamental frequency ($F0$) and intensity, along with a measure of vowel duration, were studied in 11 Broca, 7 Wernicke, 15 Parkinson, and 15 normal-control subjects performing four speech tasks: reading, counting, conversation, and vowel prolongation. Tape-recorded speech samples were played into a microprocessor controlled speech analyzer (PM 301 Voice Identification, Inc.), where $F0$ in Hertz and intensity in relative decibels were calculated. Measures of mean $F0$ and $F0$ variability, mean intensity and intensity variability, and duration of the prolonged vowel were analyzed. Values differed across speech tasks: Overall, $F0$ and intensity values were higher in reading than in the other three tasks. Further, a significant group by task interaction was found for mean $F0$. Broca and Parkinson patient groups had abnormally high $F0$ mean values, but differed from each other in $F0$ variability (Broca having high and Parkinson having low $F0$ variability) and mean intensity; surprisingly, the Broca group had intensity values significantly lower than the other groups. All three clinical groups differed significantly from normals on the duration measure. Examination of PET and CT-scan data in individual patients indicated an association between prosodic impairments and specific frontosubcortical dysfunction, areas subserving motor abilities in speech production.

BB12. Control of Japanese pitch accent by electrolarynx talkers. Noriko Kobayashi (ATR Auditory and Visual Perception Research Laboratories, Twin 21 MID Tower, 2-1-61 Shiromi, Higashi-ku, Osaka, 540 Japan)

Japanese word accent is known to be characterized by changes in vocal pitch. For speech with an electrolarynx of transcervical type, production of pitch accent is a difficult speech task, since most electrolarynges are not designed to facilitate quick pitch changes. However, skilled Japanese electrolarynx talkers manage to produce contrasts of word accent to a certain extent. This paper reports some perceptual cues and their effects for word accent produced by Japanese electrolarynx talkers. Subjects are four male laryngectomees who use electrolarynges with good speech intelligibility. Audio recordings were made during multiple repetitions of ten types of word pairs. Each pair consisted of two-mora words with identical phoneme sequences and contrasting accent types. Acoustic analyses revealed that the talkers successfully produced accent contrasts mainly by changing the segmental duration. Discussion will be made on some speech strategies that are not used by normal talkers for producing major perceptual cues but which are sometimes effective for the speech handicapped population for whom normal cues are not available.

BB13. Noise measurements of adult men's and women's voices. Susan Nittrouer (Boys Town National Institute, 555 N. 30th Street, Omaha, NE 68131), Richard S. McGowan (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511), and Donna Beehler (Boys Town National Institute, 555 N. 30th Street, Omaha NE 68131)

Two algorithms that were designed to detect noise in voiced speech were applied to the middle portion of vowels spoken by men and women in running speech. It was found that the women had as high a signal-to-noise ratio as the men, indicating that the women did not produce more turbulence noise than the men. Also, the ratio of the amplitude of the fundamental to first harmonic (the H1–H2 ratio) was found to be higher for the women than the men. This ratio is a measure of breathiness, which is sensitive to the abruptness of vocal fold closure [C. Bickley, MIT Working Papers, Vol. 1]. The evidence here indicates that turbulence noise is independent of the H1–H2 ratio and the abruptness of fold closure to some degree. The absolute jitter was found to decrease with frequency, corroborating the findings of others. Comparisons for the effect of time of day showed that jitter increased with time of day. [Work supported by NIH grants to Boys Town National Institute and NIH Grants NS-13870, NS-13617, and HD-1994 to Haskins Laboratories.]

BB14. Perturbation functions and measures: A theoretical study. Neal B. Pinto and Ingo R. Titze (Recording and Research Center, The Denver Center for the Performing Arts, Denver, CO 80204)

Acoustic analyses of the human voice have been developed to assist in clinical screening of laryngeal pathology. It is hypothesized that perturbation functions derived from parameters measured from the acoustic signal might reflect phonational irregularities due to randomness in the action potentials governing muscle tension, asymmetry in vocal fold geometry, randomness in the air stream from the glottis, interactions between source and tract, and other, as yet unknown, factors. A set of measures computed from these perturbation functions could provide a quantitative descrip-

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tion of vocal fold pathology. Unfortunately, many perturbation measures in use today were defined in an *ad hoc* fashion, with little attention paid to comparatively assessing proposed measures against existing measures. This paper represents an attempt to put the study of voice perturbations on a sounder theoretical footing. Utilizing theories of finite differences and subsampling, algorithms have been developed to aid in discriminating between perturbations based on duration of ocurrence as well as the type of perturbation, i.e., random or periodic. The measures obtained are independent, one reflecting short-term periodic perturbations, another short-term noiselike perturbations, etc. By theoretical and experimental work, we hope eventually to tie these measures in to conditions at the glottis. [Work supported by NIH.]

BB15. Analysis of hoarse voice applied to the diagnosis of laryngeal disease. Takuya Koizumi and Shuji Taniguchi (Department of Information Science, Fukui University, 3-9-1 Bunkyo, Fukui, 910 Japan)

A hoarse voice seems to have some acoustic properties that are associated with, and thus serve to indicate, the cause of hoarseness, that is, a pathological condition of the larynx. Hence it appears to be possible to determine, to some extent, the nature of laryngeal disease through an analysis of the hoarse voice caused by it. This work is concerned with the diagnosis of laryngeal disease by a hoarse voice analysis. This procedure of diagnosis is based on techniques of feature extraction and categorization in pattern recognition. Given samples of hoarse voices, a set of characteristic features is extracted from the samples by applying an appropriate analysis to them. The observed features expressed in the form of pattern points in a multidimensional pattern space are then classified into several different pattern classes, each of which represents a laryngeal disease. Among several methods of analysis that have been tried, two specific methods including ordinary spectral analysis by FFT have been found to be useful in this particular classification, or diagnosis, scheme.

BB16. Discriminant analysis of voices with and without hoarseness. Yoshinobu Kikuchi and Hideki Kasuya (Faculty of Engineering, Utsunomiya University, 2753 Ishii-machi, Utsunomiya, 321 Japan)

In order to realize an objective method for evaluating hoarse voice, the relationships between various acoustic parameters and hoarseness grade as judged perceptually by a laryngologist were analyzed. The acoustic parameters included perturbations in pitch period and peak amplitude, the intensity of the turbulent noise produced at the glottis, and the intensity level difference between the high- and low-frequency regions. These parameters were measured from vowel samples uttered by more than 2000 speakers and were subjected to the correlation and discriminant analyses. It has been found that the relative level of vocal noise in a selected frequency band agrees best with perceptual judgments. Furthermore, an error rate of 16% was obtained in discriminating between normal and hoarse voices using all of the above parameters. With carefully recorded vowel samples, however, the error rate was reduced to only 8%.

BB17. Oral and glottal gestures and acoustics of underlying /t/ in English. Sharon Y. Manuel (Room 36-529, Massachusetts Institute of Technology, Cambridge, MA 02139) and Eric Vatikiotis-Bateson (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511)

In English, an underlying /t/ can be produced in various ways. The glottal gestures may involve glottal attack (glottal stop), spreading, or voicing. The oral gestures may include complete or incomplete alveolar closure; or, in the case of place assimilation (e.g., "notebook" surfacing as [nowpbUk]), they may entail another oral gesture. Thus some surface variation is due to the particular gestures used. However, since the relative timing of glottal and oral gestures may effectively preclude acoustic realization of some of them, the acoustic record alone may not reveal just which gestures have actually been produced, and in what order. Therefore, some perceived variation may be due to differences in timing and relative strength of articulatory gestures, rather than changes in the *set* of gestures. This investigation of alveolar stops of two talkers focuses in particular on the occurrence of glottalized /t/ and shows differences in articulatory patterning depending on talker, speaking style, and phonetic context. Data are based on electropalatography, transglottal illumination, fiberoptic video of the larynx, oral air pressure, and acoustic measures. [Work supported by NINCDS grants to MIT and to Haskins Laboratories.]

BB18. Release mechanisms for stop and nasal consonants. Kenneth N. Stevens (Research Laboratory of Electronics and Department of Electrical Engineering and Computer Science, Massachusetts Institute of Technology, Cambridge, MA 02139)

Theoretical models have been developed to account for the rapidly changing acoustic events at the release of stop and nasal consonants with different places of articulation. The model for stop consonants predicts the level and spectrum of the initial transient, the burst of noise, aspiration noise, and formant transitions, and shows differences in the contribution of each of these components over the initial 10–20 ms for labial, alveolar, and velar places of articulation. For nasal consonants, the abrupt change in spectrum at the release can be explained by rapid movement of the zeros of the transfer function of the oral–nasal system, and by a somewhat slower movement of the poles. The pattern of movement of the zeros and hence the nature of the spectrum change depend on the consonantal place of articulation. Spectra data for spoken stop and nasal consonants, obtained with fine time resolution, show general agreement with the theoretical predictions, although the lack of quantitative data on dynamic and kinematic articulatory parameters makes detailed comparison difficult. [Work supported in part by a grant from NINCDS.]

BB19. Cross-language effects of vowels on consonant onsets. P. A. Keating and A. C. Cohn (Linguistics, UCLA, Los Angeles, CA 90024-1543)

Languages appear to differ in how much consonants coarticulate with vowels. Previously reported was an attempt to develop a metric of vowel effects on consonants so that quantitative comparisons of languages can be made. For each place of articulation in a given language, onset spectra are made of stop consonants before an /a/ vowel. Templates for each place of articulation are used to characterize these spectra. Then the templates for a language are tested against the other vowel contexts within that language. The extent to which the templates fail to generalize across vowels within each place of articulation is argued to reflect the degree of coarticulation in the language. An earlier study of Kana and Russian has been expanded to include English and Polish, so that four languages can be compared. As expected, the measure of template generalization (degree of coarticulation) varies across languages, though languages also differ in the behavior of individual consonants and vowels. Possible phonological motivations for such variation will be discussed. On the basis of the four languages analyzed, the advantages and limitations of using templates to construct a coarticulation metric will also be discussed. [Work supported by NSF.]

BB20. A cross-linguistic contrast in consonant and vowel timing. Caroline L. Smith (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511)

Two opposing explanations of the timing of consonants and vowels are that (1) they are timed relative to each other, or (2) vowels are produced in a continuous rhythm with consonants superimposed on them [Fowler, J. Exp. Psychol. **112**, 386–412 (1983)]. Languages with different temporal properties, such as Japanese, a mora-timed language, and Italian, a syllable-timed language, might be expected to exemplify these different relationships. Speakers of each of these languages were recorded producing pairs of disyllabic words with medial consonants of contrasting length (single or geminate). Evidence from measures of acoustic duration and vowel-to-vowel coarticulation, shown by contextual changes in $F2$, suggest that Fowler's model (2) predicts the timing patterns found in Italian, while the other model (1) predicts the behavior of Japanese. [Work supported by NSF.]

BB21. How coarse is glottal/oral coordination? John Kingston (DMLL, Morrill Hall, Cornell University, Ithaca, NY 14853-4701)

Chasaide and Gobl [J. Acoust. Soc. Am. Suppl. 1 **82**, S116 (1987)] show that the glottal opening in a voiceless stop is anticipated in a preceding vowel for some speakers of Swedish and English. This opening is only partial and the glottis is not fully abducted until the stop closure is made. Chasaide and Gobl argue that this early glottal opening and the implied coarse coordination with the oral gestures are a typical feature of the [voice] contrast in languages where the contrast is realized as unaspirated versus aspirated (they also showed that in French, where the [voice] contrast is realized as voiced versus voiceless, glottal abduction is not anticipated in the vowel preceding a voiceless stop). Data from two speakers of Icelandic [J. Kingston, J. Acoust. Soc. Am. Suppl. 1 **82**, S114 (1987)] revealed a contrast between abduction that begins in the preceding vowel, and abduction whose onset is synchronized with the oral closure in that language. Early abduction is found in Icelandic's preaspirates and abduction is synchronized with closure in other stop types [M. Petursson, Phonetica **33**, 169–198 (1976)]. Two accounts are possible: (1) following Chasaide and Gobl, the extent of overlap between glottal abduction and the preceding vowel could simply be proportional to the size of the abduction—larger abduction in the preaspirates producing more overlap with the preceding vowel; or (2) Icelandic speakers may control when abduction begins independent of how large a glottal opening is made. That the language bases a contrast on whether glottal abduction overlaps with the preceding vowel, together with other acoustic evidence favors the second alternative. [Work supported by the Arts College, Cornell University.]

BB22. Cycle to cycle spectral perturbations in the voices of male and female speakers. A. Yonovitz (The University of Texas, Speech and Hearing Institute, 1343 Moursund, Houston, TX 77030) and L. Yonovitz (Speech, Language and Learning Center, 12841 Jones Road, Houston, TX 77070)

The glottal function was obtained through the use of a reflectionless tube. This tube acted as a pseudoinfinite termination of the vocal tract. Male and female speakers phonated a neutral vowel while target matching to a 125- and a 210-Hz tone, respectively. The glottal waveform was digitized at 40*k* samples/s and each cycle was partitioned at zero crossings. This procedure provided an accurate determination of the period of each glottal pulse. Discrete Fourier analysis on each individual cycle indicated changes in the spectral content. The amplitude and phase of the first ten harmonics were utilized to obtain separate perturbation values as well as an overall amplitude measure. The data showed a systematic change in the spectral character of the sustained vowel source signal. Spectral perturbation analaysis may be a more inclusive measure of waveshape changes than jitter or shimmer.

BB23. Voice source variations during consonant–vowel transitions. Anders Löfqvist (Department of Logopedics and Phoniatrics, University Hospital, S-221 85 Lund, Sweden and Haskins Laboratories, New Haven, CT 06511) and Richard S. McGowan (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511-6695)

Studies of the voice source during speech have been concerned mostly with variations in fundamental frequency and amplitude. However, the source variations also comprise the spectral characteristics and the harmonics-to-noise ratio. The present study reports detailed measurements of voice source properties during transitions between consonants and vowels. Oral air flow, oral air pressure, and transillumination were recorded from two subjects producing reiterant speech with the vowel /æ/ and different voiced and voiceless consonants; the flow signal was inverse-filtered to obtain an estimate of the glottal pulse. Results indicate a breathy phonation at the release of voiceless consonants as indicated by an open quotient close to 1. Peak flow during each glottal pulse is thus high at voicing onset following a voiceless consonant and decreases, with an undershoot, during the first 20 pitch periods. The source pulse is also skewed to the left during the first pitch periods following voiceless consonants. Source variations following voiced consonants are less pronounced. [Work supported by NINCDS Grants NS-13617 to Haskins Laboratories.]

WEDNESDAY MORNING, 16 NOVEMBER 1988

HONOLULU/KAHUKU ROOM, 8:00 TO 11:29 A.M.

Session CC. Structural Acoustics and Vibration III and Underwater Acoustics III: Wavenumber Array Signal Processing

Albert J. Tucker, Cochairman
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Ieharu Kaihoh, Cochairman
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Chairman's Introduction—8:00

Invited Papers

8:05

CC1. Wave-vector-frequency spectral estimation: A review of conventional signal processing techniques. Wayne A. Strawderman (Naval Underwater Systems Center, New London, CT 06320)

Signal processing techniques for estimating wave-vector-frequency spectra of space-time fields from measured data parallel those developed by electrical engineers for estimating frequency spectra from measured samples of temporal fields. This paper reviews a conventional technique for estimating the wave-vector-frequency spectrum from samples of a space-time field measured through an array of sensors. The mean value of this specral estimate can be related to the true spectrum of the space-time field and the wave-vector and frequency responses associated with the various components of the signal processing. This relationship permits, under certain conditions, the quality of the spectral estimate to be assessed. The dependence of the quality of the spectral estimate on the various components of the signal processing is discussed.

8:25

CC2. An overview of array processing methods for ω–k analysis of space-time signal fields. Arthur B. Baggeroer (Departments of Ocean and Electrical Engineering, Massachusetts Institute of Technology, Cambridge, MA 02139)

Signals from an array of sensors can be used to estimate the frequency-wavenumber content of a signal field using several methods. The array geometry has a significant impact upon the signal processing methods used. In those applications where the arrays have a lattice spacing, 2-D or 3-D extensions of Fourier methods can readily be used; however, when the array has few sensors and/or sparse and irregular spacing, alternative algorithms must be used. These include approaches based upon minimum variance filters (maximum likelihood), prediction-error filters (maximum entropy), and eigenvector (MUSIC) decompositions. Many structural acoustics experiments involve broadband, transient signals, so the time variation of the wavenumber content of the field becomes important, e.g., when group as well as phase speeds are desired. There are a number of methods for transient analysis that are commonly referred to as velocity, or slowness, time spectra. These also must be tailored to the array geometry. This presentation will give an overview of the methods used for ω–k analysis with applications drawn from structural, borehole, and underwater acoustics.

8:45

CC3. Prospects for studying and interpreting wave processes induced by inhomogeneities in shells using wave-vector filtering. Allan D. Pierce (Graduate Program in Acoustics and Department of Mechanical Engineering, Pennsylvania State University, P. O. Box 30, State College, PA 16804)

Wave-vector filtering allows one to isolate particular wave processes *in situ* when a complex structure is in a random and intricate state of vibration. The probability of success and the scientific and engineering values of such an achievement strongly depend, however, on a prior knowledge of what processes are likely and of the underlying physical principles. The present paper discusses processes observable on inhomogeneous structures that, for the most part, resemble thin-walled cylindrical elastic shells. In the absence of inhomogeneities, the basic shell equations are wave-field equations for a two-dimensional anisotropic and dispersive medium. At the frequencies of interest, longitudinal, shear (transverse), and flexural wave-types are all slightly modified by the finite radius of curvature of the shell. Inhomogeneities are characterized by finite regions where the shell's physical properties differ significantly from the "ambient," or by lumped parameter quantities (such as added mass) at discrete points or along lines. In general, an incoming wave of any one type at an inhomogeneity will excite outgoing waves of all three types. At line inhomogeneities, the outgoing waves will correspond to a version of the trace velocity matching principle. Wave-vector filtering measurements in conjunction with a knowledge of the physics underlying the wave processes should allow one to indirectly infer gross properties of inhomogeneities. Energy accountancy principles, for example, enable one to determine the energy dissipation rate at an inhomogeneity. [Work supported by ONR.]

9:05

CC4. The application of multichannel measurement techniques in structural acoustics. D. Feit (ONR London Branch Office, FPO New York, NY 09510)

Measurements made on distributed arrays of sensors can be used to extract useful characteristics of the structural acoustic fields and the sources generating them. In this presentation, two experiments performed on elastic cylindrical shells immersed in water are discussed. In one case, the shell could be considered to be unstiffened, while, in the other, the shell had discrete ring stiffeners. Measurements were performed with both line arrays of accelerometers as well as hydrophones. Wavenumber frequency plots are obtained in all cases. This type of analysis on the accelerometer data shows clear evidence of both quasiflexural as well as quasilongitudinal waves in the shell response. The hydrophone data indicate the presence of two strong acoustic components, one traveling at grazing incidence to the array, while the other passes over the hydrophone array with a trace wave speed coincident with a wave generated on the shell traveling with the speed characteristic of a quasilongitudinal wave. For the measurements performed with the rib-stiffened shell, the wavenumber frequency plots show the presence of the ribs as aliased versions of the dispersion curves for the unribbed case. The theoretical basis for this is derived.

9:25

CC5. An underwater acoustic imaging method using cross-fan beam scanning. Ieharu Kaihoh, Hisami Hayakawa, and Masanori Osanai (OKI Electric Industry Co., Ltd., 688 Ozuwa, Numazu, 410 Japan)

A real-time underwater acoustic imaging method named cross-fan beam scanning has been developed by OKI Electric Co., Ltd. In this method, acoustic projector elements are arranged vertically, and hydrophone elements are arranged horizontally to form a cross-linear array. That transmitting array generates thin horizontal fan beams, and the receiving array generates multi-thin-fan preformed vertical beams. Total beam patterns are obtained as the product of the transmission and reception beam patterns, so that narrow pencil beams are generated at the intersection of the two beams. Transmission and reception beams are electrically scanned, respectively, and the resulting pencil beams can be pointed at in any direction in a moment. Using this method, several types of imaging systems were completed. These are installed in submersibles, surveying ships, and work vessels such as the "Shinkai-6500" deep-sea manned submersible of the Japan Marine Science and Technology Center and the "Marcas-2500" deep-sea ROV of KDD, etc. In this paper, the imaging principle, the sonar system design, and the characteristics of typical systems are described and the imaging results are shown.

9:45

CC6. Wave-vector filtering of holographic data on point-driven cylinders. Karl Grosh (Sachs/Freeman Associates, Inc., 1401 McCormick Drive, Landover, MD 20784) and Earl G. Williams (Naval Research Laboratory, Code 5137, Washington, DC 20375-5000)

Generalized nearfield acoustical holography was used to provide the radial velocity $v(z,\phi,\omega)$ at over 8000 locations on a complete cylindrical surface tangent to a point-driven, submerged capped cylinder. Because of circumferential periodicity, circumferential waves can be extracted by the use of simple FFT methods, yielding $v(z,n,\omega)e^{in\phi}$, where n is the circumferential wavenumber. To obtain an axial decomposition of the waves, the FFT fails due to insufficient resolution and spectral leakage. Instead a high-resolution method is used, the extended Prony method. Key to the success of this method is the use of singular value decomposition to estimate the number of waves (model order) present at a particular frequency and value of n. With the model-order estimation, the Prony method provides accurate location of the poles corresponding to the desired wavenumbers. An iterative scheme is used to refine the whole process. The resulting waves can be related to the fluid-loaded modes of the driven structure.

Contributed Papers

10:05

CC7. Spectral parametrization of wave scattering from a rigid aperture coupled enclosure with interior loading. L. B. Felsen (Department of Electrical Engineering and Computer Science/Weber Research Institute, Polytechnic University, Farmingdale, NY 11735) and G. Vecchi (CESPA, Politecnico, Torino, Italy)

Wave interaction with complex structural environments can be analyzed by regarding the overall response as the response from coupled subsystems. The decomposition into subsystems is wavenumber-scale dependent, and the "best" classification involves weak coupling between subsystems but strong coupling of wave types within a subsystem. How to effect the classification is illustrated here for a cylindrical thin smoothly convex rigid shell, which grants access to an enclosed cylindrical convex rigid interior load via a narrow-slit aperture. The geometry suggests a parametrization of exterior wave phenomena by geometrically reflected, slit diffracted, and surface guided creeping waves, and of interior wave phenomena by guided local modes in the nonuniform waveguide formed by the region between the shell and interior boundary. The slit produces coupling among the exterior wave types and among the interior wave types, as well as between the exterior and interior. These couplings cause spatial wavenumber conversions from trapped, to radiating, etc. A system of state vectors, propagation matrices, and coupling matrices renders the overall description self-consistent, and permits the development of reduced forms by loading a significant (for example, resonant) subsystem with nonresonant remainders. The dynamics of the parametrization is illustrated by considering the numerical example of plane wave scattering as a function of aspect and frequency when the interior and exterior boundaries are eccentric circular cylinders. [Work supported by ONR and David Taylor NRDC.]

10:17

CC8. Examples of two wavenumber spectra in nonhomogeneous one-dimensional structures. Robert M. Kennedy (Naval Underwater Systems Center, AUTEC Detachment, West Palm Beach, FL 33402-7517) and Wayne A. Strawderman (Naval Underwater Systems Center, New London, CT 06320-5594)

Structures of interest to hydroacoustic research frequently have spatially varying properties that cause the field variable to be statistically nonhomogeneous. Two wavenumber spectra of one-dimensional nonhomogeneous fields have been defined [W. A. Strawderman, J. Acoust. Soc. Am. Suppl. 1 80, S15 (1986)], but there exists little experience in interpreting these spectra. This paper presents a first look at the two wavenumber spectra of various theoretical and measured examples of nonhomogeneous fields. The examples are chosen to form a progression of two wavenumber spectra from simple canonical forms of exponentially damped traveling waves to more complex measured nonhomogeneous fields on axisymmetric periodic structures. The desired two wavenumber spectra are obtained from calculated or measured cross-specral density matrices by use of an efficient algorithm. A comparison of the spectra of both the canonical and measured fields displays a common structure observed in both. One and two wavenumber spectra of measured data are compared to illustrate the potential simplification and interpretive role of two wavenumber spectra.

10:29

CC9. Time varying wavenumber-frequency spectra. Joseph A. Clark and David Feit (David Taylor Research Center, Washington, DC 20084)

Under conditions of transient excitation, displays of time varying spectra can reveal significant space-time features of structural response as well as stationary wavenumber-frequency features. An example obtained from an investigation of wave propagation on a ribbed, cylindrical steel structure will be presented in this talk. The structure was immersed in water, and a hydrophone array was employed to measure responses to impulsive loading at 4096 points on a surface around the structure. A series of wavenumber-frequency plots were generated by transform processing selected temporal portions of the time-space data. The transformed data can be viewed either as veritcal layers of 2-D surfaces or as a motion picture of a single wavenumber/frequency surface. The displays of time varying spectra have been found useful as tools for analyzing the growth and decay of modes propagating along the structure as well as the coupling between modes.

10:41

CC10. Wavenumber-frequency calibration for planar arrays. D. H. Trivett, L. D. Luker, S. Petrie, and A. L. Van Buren (Naval Research Laboratory, Underwater Sound Reference Detachment, P. O. Box 568337, Orlando, FL 32856-8337)

Wavenumber-frequency calibration of planar receiving arrays requires the ability to generate homogeneous, single wavenumber pressure fields over the surface of the array. When the wavenumber-frequency region of interest is evanescent, transmitting arrays previously designed to generate such fields have been found to generate fields contaminated with harmonics, acoustic wavenumbers, and nonacoustic wavenumbers from the excitation of antisymmetric Lamb waves. Two transducers that greatly reduce contamination have recently been constructed using sheets of PVDF with 20 independent rectangular electrode stripes. The stripes are driven with shading coefficients generated by a simple point source numerical algorithm and are found to be capable of generating homogeneous, single wavenumber, evanescent pressure fields over a specified region. The two arrays differ in their method of eliminating contamination due to the excitation of antisymmetric Lamb waves. One array shifts the phase velocity of the Lamb wave out of the region of interest by bonding the PVDF directly to a thick plate of Lexan. The second array decouples the PVDF laterally by the use of slicing along the electrode stripes and plotting in a heavily damped elastomeric material. Measured pressure fields and phase velocities of both arrays are compared with numerical calculations. [Work partially supported by ONT.]

10:53

CC11. A wavenumber domain analysis of the coupling of a structural mode and flow turbulence. Y. F. Hwang and G. Maidanik (David Taylor Research Center, Bethesda, MD 20084-5000)

Previous studies on the coupling of a structural mode and the pressure field induced by a turbulent boundary layer have concentrated on the determination of the relative contributions of two regions of the wavenumber domain; the high-wavenumber region at and near hydrodynamic coincidence and the low-wavenumber region at and near the structural free wavenumber. The corresponding contribution to the coupling by the intermediate-wavenumber region that lies between the high- and low-wavenumber regions has not received proper attention. In this paper, formulas to estimate the contributions to the coupling by the three wavenumber regions are derived, and numerical results are evaluated and compared. It is found that situations exist in which the contribution by the intermediate-wavenumber region may be significant. [Work supported by ONR.]

11:05

CC12. Statistical reliability of wavenumber spectrum estimates with crossed-line multiplicative arrays. Frank Berkman, Nathan Martin, and Dale Korff[a] (BBN Systems and Technologies Inc., 10 Moulton Street, Cambridge, MA 02238)

In a previous paper [J. Acoust. Soc. Am. Suppl. 1 **64**, S124, (1978)], it was shown that in a spatially homogeneous noise field the mean response of a multiplicatively processed pair of crossed-line arrays with apertures $2L_1$ and $2L_2$ and appropriate shading is identical to the mean-squared response of a rectangular, separably shaded planar array with aperture $L_1 \times L_2$. Thus the number of elements required for wavenumber spectrum measurement may be considerably reduced (from N^2 to $4N$ for a square array) or the wavenumber resolution may be significantly increased for the same number of elements by the use of multiplicative crossed-line arrays. However, there is an additional trade-off of statistical reliability of the measurement for finite averaging time. This paper compares the variance of the wavenumber spectrum estimate obtained with rectangular planar arrays and with crossed-line multiplicative arrays. [a] Presently at Raytheon Submarine Signal Division, Portsmouth, RI.

11:17

CC13. Role of spectral derivatives of structure reactance in vibration problems. K. L. Chandiramani (Bolt Beranek and Newman Inc., 10 Moulton Street, Cambridge, MA 02238)

Energy density and energy flux of resonant vibration of fluid-loaded structures are expressed in terms of spectral derivatives (i.e., derivatives with respect to frequency ω and wavenumber k) of structure surface wave reactance X. Two methods of derivation are presented: (1) Whitham's variational method is used after time-averaged Lagrangian density is expressed in terms of the structure reactance X; (2) Pole contributions to structure transfer line admittance are evaluated. For small dissipation, both methods yield identical formulas for spatial and temporal decay rates of resonant vibration. Both methods are also capable of extensions that enable consideration of instabilities resulting from the presence of a uniformly moving fluid adjacent to the structure. [Work supported by the Office of Naval Research.]

Session DD. Underwater Acoustics IV: Propagation in Underwater Acoustics (Precis-Poster Session)

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Contributed Papers

Following presentation of the précis, posters will be on display until 11:30 a.m.

8:00

DD1. Efficient computation of ideal starting fields for parabolic equation models. E. Richard Robinson and David H. Wood (Code 3122, New London Laboratory, Naval Underwater Systems Center, New London, CT 06320)

When one replaces the Helmholtz equation by the related parabolic equation, the resulting gain is the relative numerical ease of solving the parabolic equation using a marching type algorithm. However, one is left with the problem of providing the initial data—or the starting field for the PE model. Although it is well known that the normal mode starter is the corrected initial data, it is not widely used because of the expense involved in the compuation. In this approach, however, the idea that one need not compute the normal modes individually to obtain the correct starting field is exploited. If one considers the function represented by a weighted sum of the normal modes, then it is found that the correct starting field can be obtained by computing the $-\frac{1}{4}$ power of the matrix generated from the FFT of the index of refraction. This result is derived and some examples of the starting fields are presented and compared to starters computed or approximated in other ways. Further, the effects on PE model outputs using the various starters are demonstrated.

8:05

DD2. Modifying sound speed to reduce phase errors in parabolic equations by a Toda flow method. Kenneth R. Driessel (Department of Mathematics, Idaho State University, Pocatello, ID 83209) and David H. Wood (Department of Computer and Information Sciences, University of Delaware, Newark, DE 19716)

It is well known that the standard parabolic wave equation, in the absence of range dependence, has the correct normal nodes (amplitudes) but incorrect eigenvalues (phase). This is regrettable because the desired information is generally more sensitive to phase errors than to amplitude errors. Several authors have attempted to *increase* the accuracy of the phases by modifying the sound-speed profile, even though this decreases the accuracy of the amplitudes. Attempts have been made to *eliminate* the phase errors (accepting the increase in amplitude errors) by a Toda flow method. From a given sound-speed profile, one can easily generate a matrix with the desired eigenvalues. The spectral mapping theorem shows that a certain square root of this matrix would have eigenvalues that are distorted in the *opposite* way from the parabolic wave equation. The modified sound-speed profile that corresponds to the last matrix is sought: A system of spectrum-preserving differential equations is solved giving the evolution of this matrix into one where the modified sound-speed profile can be recovered.

8:10

DD3. Speeding up the IFD model by high-order numerical methods. Ding Lee (Naval Underwater Systems Center, New London, CT 06320) and Faisal Saied (Department of Computer Science, Yale University, New Haven, CT 06520)

The implicit finite difference (IFD) model implements an IFD scheme of the Crank–Nicolson type to accurately solve a wide angle wave equation. To maintain the desired accuracy, in some cases, the IFD requires small step sizes in both depth and range directions, which increases the computation time. A number of high-order finite difference methods for solving equations of the Schrödinger type have been developed. These high-order methods are not only accurate but also allow the use of larger step sizes in both depth and range directions. A fourth-order implicit finite scheme (IFD4) is used to solve the parabolic wave equation as a demonstration of the speed advantage. While accuracy is maintained, the IFD4 offers much faster speed than the Crank–Nicolson IFD model. Numerical results will be presented to show the savings in computing time achieved by these methods. Theoretical developments and computational aspects will be discussed. [Work supported jointly by NUSC and ONR.]

8:15

DD4. Computational results using FOR3D and an analytic eddy model. P. D. Scully-Power, D. Lee, G. Botseas (Naval Underwater Systems Center, New London, CT 06320), and W. L. Siegmann (Rensselaer Polytechnic Institute, Troy, NY 12180)

A three-dimensional parabolic approximation model FOR3D has been developed. Its validity has been verified by means of test examples and comparisons. The present implementation of the model has some desirable capabilities, and additional useful ones are under development. A primary advantage of the model is the availability of treating azimuthal coupling. The FOR3D has been used to process environmental data obtained from ocean prediction models. In this report, an FOR3D is used to examine acoustic propagation through a parametric eddy model. Computational results will be presented, and issues relating to three-dimensional effects will be discussed. [Work supported jointly by NUSC and ONR.]

8:20

DD5. A marching-frame, finite-element, range-dependent, ocean acoustic propagation and scattering model for long-range acoustic propagation simulation. Joseph E. Murphy (Department of Physics, University of New Orleans, New Orleans, LA 70148) and Stanley A. Chin-Bing (Naval Ocean Research and Development Activity, Numerical Modeling Division, Stennis Space Center, MS 39529-5004)

The ocean acoustic propagation and scattering simulation model based on the finite-element method [J. E. Murphy and S. A. Chin-Bing, Math. Comput. Modeling **11**, 70–74 (1988)] has been extended to long ranges using a marching-frame approach. This Finite-element, Full-wave Range-dependent, Acoustic Marching Element (FFRAME) model has the advantage of a forward marching model (similar to the marching algorithms of parabolic equation models) while allowing that within each elemental frame the computed pressure field is full wave and range dependent. Successive frames are coupled by imposing the pressures from the right-hand side of the Nth frame onto the corresponding nodes on the left-hand side of the $(N+1)$th frame. The size of each elemental frame can be adjusted to include all of the significant effects due to full-wave propagation and scattering within that frame. Thus the forward propagating field between elemental frames correctly includes the losses due to backscatter. Numerical examples are presented that illustrate the advantage of the FFRAME model over conventional one-way ocean acoustic propagation models. [Work supported by ONR and NORDA.]

8:25

DD6. Long-range, range-dependent, acoustic propagation simulation using a full-wave, finite-element model coupled with a one-way parabolic equation model. Stanley A. Chin-Bing (Naval Ocean Research and Development Activity, Numerical Modeling Division, Stennis Space Center, MS 39529-5004) and Joseph E. Murphy (Department of Physics, University of New Orleans, New Orleans, LA 70148)

A full-wave, range-dependent, ocean acoustic propagation and scattering model based on the finite-element (FE) method [J. E. Murphy and S. A. Chin-Bing, Math. Comput. Modeling **11**, 70–74 (1988)] has been combined with a range-dependent, one-way parabolic equation (PE) model to simulate long-range (>100 km) ocean acoustic propagation. This hybrid FE/PE model has the advantage that it includes the coupled effects of the full-wave, range-dependent, forward propagated and backscattered fields where full-wave effects are important, while allowing for the much faster computational marching method of the PE model in those ocean regions where full-wave effects are not significant. The FE/PE hybrid model also uses the FE full-wave, range-dependent solution from the source out to a specified range at which the pressure field serves as the initial field input to the PE model. This PE starting field is most useful when the source is near a severely range-dependent boundary. Numerical examples are given and comparisons made with the more "exact" coupled mode and finite-element models that illustrate the increased accuracy of this hybrid FE/PE model over the one-way propagation models. [Work supported by ONR and NORDA.]

8:30

DD7. Acoustic wave propagation employing Gaussian wave packets. F. J. Ryan (Code 541, Naval Ocean Systems Center, San Diego, CA 92152-5000)

Recently, there has been much interest in using Gaussian beam methods to solve wave propagation problems in spatially inhomogeneous media. A new full-wave propagation technique, multiple Gaussian wave packets (MGWP), will be described, which combines some features of Gaussian beam methods and the split-step PE algorithm. Briefly, the MGWP technique consists of expanding the initial wave field at some starting range into a series of complex Gaussian wave packets and then allowing the wave packets to evolve with range. Each wave packet has the functional form of a Gaussian with complex parameters that control the phase-space distribution of the wave packet. The range evolution of the Gaussian parameters is governed by a Schrödinger equation and is exact for refrative index profiles that are quadratic. Comparisons of the MGWP method with a normal mode propagation model will be shown. [Work supported by NOSC IR.]

8:35

DD8. Weighted-Gaussian-quadrature-generated irrational functions for ray trajectories for range-independent and range-dependent problems. Edward R. Floyd (Arctic Submarine Laboratory, Naval Ocean Systems Center, San Diego, CA 92152-5000)

A parametric description of ray trajectories in terms of irrational functions was developed. These irrational functions are generated by applying weighted-Gaussian quadratures (WGQ) to a parametric Hamilton–Jacobi representation of the ray trajectory. The irrational function may be developed in principle to any order as it is consistent with the order of our weighted-Gaussian quadrature. The Hamilton–Jacobi transformed canonical variables are constants for the range-independent problem but are functions of range for the range-dependent problem. The range dependence of the transformed canonical variables has been developed by nonadiabatic range-dependent canonical perturbation theory. The irrational function description of the ray facilitates a description of the perturbing Hamiltonian in terms of only range and the transformed canonical variables. The perturbation expansion of the transformed canonical variables is structured to be consistent with the expected WGQ remainder. This perturbation technique is applicable to cases whose unperturbed Hamiltonians cannot be integrated in closed form.

8:40

DD9. Complex ray methods for flat and sloping waveguides. Evan K. Westwood (Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78713-8029)

The problem of the interaction of a spherical wave at a plane, penetrable interface is solved in terms of ray quantities by using the saddle point method. When the plane-wave reflection and transmission coefficients are allowed to influence the saddle point criterion, the saddle points become complex, in general. As a results, the reflected lateral wave field and the transmitted evanescent field are included in this formulation. This method for solving the single interface problem is extended in a straightforward manner to the flat waveguide (the Pekeris model) and the penetrable wedge. In both cases, the total field is expressed as a sum of ray fields. For the flat waveguide, comparisons show nearly perfect agreement between the ray model and the SAFARI model at all ranges. In contrast, the discrete normal mode solution is incorrect (1) at short ranges in all cases, and (2) at all ranges when a mode is near cutoff. For the wedge problem, comparisons between the ray model and a two-way coupled mode model show close agreement. The interesting phenomenon of multiple lateral waves in a wedge will be discussed. [Work supported by Office of Naval Research Contract N00014-87-K-0346.]

8:45

DD10. Complex source pulsed beams: New test pulses for transient acoustic propagation and diffraction modeling. E. Heyman, B. Z. Steinberg (Department of Electrical Engineering, Tel Aviv University, Tel Aviv 69978, Israel), and L. B. Felsen (Department of Electrical Engineering and Computer Science/Weber Research Institute, Polytechnic University, Farmingdale, NY 11735)

Complex source pulsed beams (CSPB) are transient waveforms generated when the space coordinates and initiation time of a pulsed line source (two dimensional) or point source (three dimensional) are assigned complex values. They generalize to the time domain the time-harmonic, complex-source-point Gaussian beams. A CSPB is described by the analytic signal obtained from analytic continuation of the conventional impulsive time-dependent, free-space Green's function to the domain of the complex source parameters that control the spatial and temporal widths of the pulsed beam. Numerical examples illustrate the CSPB shapes obtained for various choices. Generalized CSPB fields can be constructed by convolving the CSPB with various time functions. The general theory of the CSPB is discussed. It is also shown how to apply the CSPB as the excitation of a propagation or diffraction environment when the complex space-time source coordinate substitution is performed on the conventional space-time Green's function for that environment. This yields new formulations for transient propagation and diffraction. Examples include a two-fluid medium separated by a plane interface, and diffraction by a knife edge. [Work supported by ONR.]

8:50

DD11. Benchmarking range-dependent, ray-tracing codes in non-Cartesian coordinates. Sze M. Tan and Gary E. J. Bold (Physics Department, Auckland University, Private Bag, Auckland, New Zealand)

The problem of validating computer codes for numerical ray tracing in range-dependent, sound-speed profiles will be discussed. To find accurate travel times for eigenray computation, one must obtain precise solutions of the ray equations parametrized by time. A method for finding nontrivial sound-speed variations based on conformal transformations is described that allows analytic solutions from simpler ray-tracing problems to be used for computing ray paths through the resulting range-varying medium. Besides its application to validating computer codes, the method can be used to convert a ray-tracing or eigenray-finding problem originally formulated in terms of a circular or elliptical coordinate system into an associated problem on a Cartesian coordinate system. Sound-speed fields specified on the natural grid for the non-Cartesian system map directly onto the Cartesian grid, allowing the use of more computationally efficient Cartesian ray-tracing algorithms. Travel times are not affected by the transformation. Hence, the method is particularly suited to predicting arrival times associated with eigenrays. Illustrative ray traces and results will be shown.

8:55

DD12. Chaotic wave propagation in random media. Alexander Pidwerbetsky (AT&T Bell Laboratories, Whippany, NJ 07981)

Under certain conditions, wave propagation in a random medium can become chaotic, i.e., display extreme sensitivity to initial conditions and complex behavior. This problem has been treated by both ray and wave theory as well as simulations. Chaos is shown by the presence of a positive Lyapunov exponent. For some random media (irregularity spectra), the Lyapunov exponent can be theoretically calculated and agrees with the simulation results. In a ray model, a positive Lyapunov exponent results in the exponential divergence of nearby rays. In the wave case, narrow beams diverge exponentially with range, provided other effects (e.g., diffraction and scattering by small-scale irregularities) do not dominate. In both ray and wave cases, the local exponential divergence results in tangling of the global portraits of the wave fields in phase space. In regimes where it dominates, chaotic wave propagation has wide ranging implications for underwater acoustics, including the proliferation of caustics and uncertainties in predicting long-range propagation. Environments that produce multiple scattering (volume, surface, or bottom) by fluctuations with a large length scale are most likely to show chaotic wave propagation.

9:00

DD13. New techniques for investigating the properties of chaotic ray paths. D. R. Palmer (NOAA/AOML, 4301 Rickenbacker Causeway, Miami, FL 33149), T. M. Georges, and R. M. Jones (NOAA/WPL, 325 Broadway, Boulder, CO 80303)

It has been established that acoustic ray paths in a range-dependent ocean environment can exhibit chaotic behavior [Palmer *et al.*, Geophys. Res. Lett. **15**, 569–572 (1988)]. The usual techniques for identifying chaotic rays are the examination of Poincaré sections and power spectra of path depth, as well as the observation of exponential sensitivity to initial conditions. These techniques are not always useful, however, and are not directly related to observable signal characteristics. Travel times, ray elevation angle at axis crossings, and upper and lower turning point depths have practical relevance and provide new insights into the character of chaotic rays. Since this effort involved the numerical calculation of ray paths for both the Helmholtz and parabolic equations, procedures were developed for comparing results obtained for the two equations.

9:05

DD14. A coupled-mode method for sound interaction with an elastic oceanic bottom. Juan I. Arvelo (Code U25, Naval Surface Warfare Center, Silver Spring, MD 20903-5000), Maryline Talmant, and Herbert Überall (Physics Department, Catholic University of America, Washington, DC 20064)

The theory of coupled modes with elastic waves from the bottom sediments of a shallow oceanic environment is developed for a source at the water column. The shear and compressional attenuation coefficients have been included as a first- and second-order perturbation to the solution of the purely real eigenequation. Results of these calculations display the importance of the second-order perturbation calculation when radiating modes are included in the transmission loss calculations or when shear waves are included. The perturbed eigenvalues and eigenfunctions were used by an already developed coupled normal-mode computer code to provide the transmission loss in a range-dependent, shallow-ocean environment [Miller *et al.*, J. Acoust. Soc. Am. **79**, 562–565 (1986)]. The results obtained are compared to the ones by Miller *et al.* for upslope and downslope environments. [Work supported in part by the Individual Research Board at the Naval Surface Warfare Cener and by NORDA.]

9:10

DD15. A long-range acoustic experiment with sampling in the internal wave fluctuation band. Timothy F. Duda, Stanley M. Flatté (Physics Department, University of California at Santa Cruz, Santa Cruz, CA 95064), Harry A. DeFerrari, and Hien B. Nguyen (Department of Applied Marine Physics, R.S.M.A.S., University of Miami, Miami, FL 33149)

Transmission at 460 Hz between two moorings centered about 600 km east of Miami is scheduled for August 1988. The 10-day, 260-km experiment has been designed to measure fluctuations induced by ocean internal waves (IW). The use of a 64-element, 150-m vertical receiver array, 6/h pulse rate, and high bandwidth allows measurement of pulse travel time (phase) to a fraction of a carrier period, and allows estimation of phase and intensity coherence times and the vertically lagged correlation function of the acoustic field, including the rms vertical wave front tilt. Path-integral solutions of the parabolic wave equation yield predictions of these statistical properties. For the Garrett–Munk IW model, rms travel-time predictions are 2–6 ms, and rms vertical angle predictions are 0.2–0.7 deg, with the lowest values for shallow turning rays and the largest for near axial rays. Expected phase coherence time scale is 5 min., with intensity time scale varying between 1 and 30 min., indicating the presence of both partially and fully saturated rays. These quantities yield a depth-dependent measure of rms vertical IW displacement, and some information about the IW spectrum.

9:15

DD16. A VLF propagation experiment off the coast of Oregon: Measurements and numerical predictions. C. A. Fisher, Hassan B. Ali, and Mona Authement (Naval Ocean Research and Development Activity, John C. Stennis Space Center, MS 39529-5004)

The Naval Research and Development Activity (NORDA) recently conducted a very low-frequency (VLF) acoustic experiment in a sloping bottom environment off the coast of Oregon. A 16-element vertical hydrophone array was deployed in water depths of 200 and 400 m. An upslope and two cross slope explosive shot lines were completed. In addition, an upslope cw (15 Hz) tow was recorded. This paper uses the results of this experiment to investigate the importance of bottom shear and bathymetric variation on energy partitioning between water and ground paths. The measured data are compared with predictions from SAFARI, a range-independent model that allows conversion to shear energy in the bottom, and PE, an acoustic model that allows variable bathymetry. Two different PE codes are used, viz., IFDPE and a PE pulse code recently developed at NORDA (M. Collins), which allows for subbottom structure that is not necessarily parallel to the ocean bottom. Measured cw propagation loss and explosive shot record sections are compared with predicted results from both SAFARI and PE.

9:20

DD17. A survey of acoustic techniques for monitoring El Niño. T. M. Georges, D. R. Palmer,[a] R. M. Jones, and J. P. Riley (NOAA Wave Propagation Laboratory, Boulder, CO 80303)

The challenge of understanding the El Niño–Southern Oscillation (ENSO) cycle in the equatorial Pacific Ocean is a test of one's abilities to observe, model, and forecast the processes of global climate change. Monitoring the structure, dynamics, and energetics of the ocean interior on the space-time scales of the ENSO signal appears to be a task for acoustic remote sensing. Therefore, the following acoustic strategies for monitoring ENSO-induced changes in the upper ocean are examined: (1) ocean acoustic tomography, (2) a long-range acoustic thermometer, (3) passive monitoring of ambient acoustic noise level, (4) an occultation technique that depends on bottom absorption, and (5) space-time scintillation analysis. Models of the ocean's acoustic properties are formulated from measurements made during the 1982–1983 ENSO event and simulated acoustic amplitude and travel-time measurements to find out how sensitive they are to the temperature changes that accompany a strong El Niño.
[a] Also at NOAA Atlantic Oceanographic and Meteorological Laboratory, Miami, FL 33149.

9:25

DD18. Modeled time variability of acoustic propagation through a Gulf Stream cold core eddy. Leonard E. Mellberg (Naval Underwater Systems Center, Newport, RI 02841-5047), Allan R. Robinson (Harvard University, Division of Applied Sciences, Cambridge, MA 02138), and George Botseas (Naval Underwater Systems Center, New London, CT 06320)

In a previous paper, A. R. Robinson [J. Acoust. Soc. Am. Suppl. 1 **82**, S2 (1987)] presented an overview and results from an ongoing forecast scheme for Gulf Stream oceanic mesoscale fields (temperature, sound speed, etc.) (gulfcasting). The coupling of the dynamical ocean model with an acoustic model and preliminary results were also presented. In the present paper, the coupled models are utilized to investigate the time variability of 25-Hz acoustic propagation through a Gulf Stream cold core eddy. Time variations of 2 days, 4 days, and 1 week are presented for ranges out to the second convergence zone. Shifts in propagation loss patterns of more than 5 km and variations in propagation loss of 5 dB are observed for a time change of 2 days. The relationships between the oceanographic and acoustic fields for the various time periods are discussed.

9:30

DD19. Three-dimensional acoustic mode propagation in the Gulf Stream. Ching-Sang Chiu and Laura L. Ehret (Institute of Naval Oceanography, Stennis Space Center, MS 39529-5005)

Based on the coupled mode theory, an efficient 3-D (i.e., three-dimensional) acoustic code for the computation of sound propagation through mesoscale ocean features was developed. The technique numerically integrates a set of first-order differential equations governing the complex envelopes that modulate, mode by mode, the adiabatic mode solution. Sensitivity studies showed that the slowly varying envelopes associated with the forward propagating components can be computed accurately using a step size much longer than the acoustic wavelength. Calculations further showed that the backscattered field is negligible even in frontal systems having extremely large sound-speed gradients. Gulf Stream sound-speed fields generated by a Harvard University ocean forecast model were used to investigate the three-dimensional environmental effects on sound propagation, both cross and along stream. In order to examine the significance of horizontal sound diffraction, the 3-D solution was compared to the N by 2-D solution. Comparison between the 3-D coupled mode and adiabatic approximation calculations was also made in an effort to quantify the validity of the adiabatic approximation.

9:35

DD20. Three-dimensional, bottom-interacting propagation in the ocean. Michael D. Collins (Naval Ocean Research and Development Activity, Stennis Space Center, MS 39529)

A three-dimensional parabolic equation (3DPE) that handles wide angles in both the vertical and the azimuth is difficult to solve numerically. This motivated the development of a 3DPE that accounts for wide angles only in the vertical [Lee *et al.*, Yale University, Department of Computer Science, Res. Rep. 463 (1986)]. An asymptotic derivation of a 3DPE that handles wide angles in the vertical will be presented as well as an efficient numerical solution based on the alternating direction implicit (ADI) method. A wide-angle starting field for initializing the 3DPE has been validated both asymptotically and numerically. Resutls will be presented for point source, multipole, and scattered fields including a benchmark calculation that demonstrates the validity of the ADI solution and the ability of the model to handle variations in azimuth. Shallow-water environments involving variable ocean depth will be considered including wedges and mounds. Two-dimensional and three-dimensional calculations will be compared.

9:40

DD21. A generalized canonical sound-speed model. Scott C. Daubin, Lan Nghiem-Phu, and Frederick D. Tappert (Daubin Systems Corporation, 104 Crandon Boulevard, Suite 400, P.O. Box 490249, Key Biscayne, FL 33149)

A generalized canonical sound-speed (GENCASS) numerical environmental model has been developed to support range-dependent underwater acoustical propagation models, especially the PE/SSF model. The GENCASS model provides an interface between the environmental data base and acoustical model; it could guide and improve the cost effectiveness of the processes of environmental data acquisition, processing, storage, and access. It extends the usefulness of Munk's canonical model to include conditions where no subsurface axis exists, as in the arctic and subarctic, and provides for such common phenomena as mixed layers, fronts, eddies, and the double minima found in the region influenced by the Mediterranean outflow. It provides a natural parametric hierarchy for organizing environmental data, based on phenomenological time scales, ranging from climatological to seasonal to diurnal. In temperate regions, four parameters are sufficient to define the model: scale depth, bottom temperature, extrapolated surface temperature, and either actual surface temperature or mixed layer depth, which are functionally related. The GENCASS model introduces a surface thermal layer depth to account for surface superheating that may occur in tropical waters. A mixing function is introduced to account for the transition through oceanographic fronts, across eddies, and through the double minima profile. A companion paper will use the GENCASS model to support PESOGEN-II illustrative runs.

9:45

DD22. Interactive 3-D canonical acoustic modeling. Lan Nghiem-Phu, Scott C. Daubin, and Frederick D. Tappert (Daubin Systems Corporation, 104 Crandon Boulevard, Suite 400, P.O. Box 490249, Key Biscayne, FL 33149)

There is often the question of whether the behavior of acoustic signals can be predicted in the complex and changing ocean environment. Our deplorable state of prediction is either blamed on limitations of acoustic models, or on the usually incomplete knowledge of the ocean environment, or on both. Despite the fact that efforts are being made to provide the critical links between the fields of oceanography and acoustics, still lacking is a comprehensive strategy to measure or predict oceanic features in order to improve our acoustical predicting capability. In a companion paper, a generalized canonical sound-speed (GENCASS) numerical environmental model is discussed. This GENCASS model is implemented in the Pesogen computer system [J. Acoust. Soc. Am. Suppl. 1 **75**, S26 (1984)] for illustration of the feasibility of using such a technique in an interactive acoustic modeling machine to better understand the interrelationships between complicated environmental conditions and their acoustical consequences. The interactive nature of this process allows rapid understanding of the database requirements, in preparation for setting environmental knowledge bounds toward better predicting capability. Examples will be shown to demonstrate the realistic nature of the canonical oceans.

9:50

DD23. A technique for obtaining acoustic levels from geophone data. C. A. Fisher (Naval Ocean Research and Development Activity, John C. Stennis Space Center, MS 39529-5004)

The hydrophone, which measures pressure, is the instrument most often used for measuring acoustic phenomena in underwater environments. A geophone, which measures particle motion, can also be used to evaluate acoustic phenomena in the ocean. In the ideal case of an acoustic plane wave, the particle motion sensed by the geophone can be converted to the equivalent pressure with an impedance correction factor. This simple approach is not appropriate in situations where the acoustic wave is nonplanar. This paper compares pressure and particle motion predicted by the SAFARI pulse model to obtain a transfer function for converting particle motion to pressure. For various scenarios, this transfer function is compared with the commonly used impedance correction factor.

9:55

DD24. The impact of a *p*H-dependent attenuation formula on previous estimates of surface loss and bottom loss. David G. Browning, Peter M. Scheifele (New London Laboratory, Naval Underwater Systems Center, New London, CT 06320), and Robert H. Mellen (Kildare Corporation, 95 Trumbull Street, New London, CT 06320)

Determination of surface loss and bottom loss from at-sea measurements requires the assumption of a propagation loss model. In most cases, this has consisted of a simple spreading loss (spherical and cylindrical) model plus attenuation. The recent development of a *p*H-dependent attenutation formula will have an impact on previously reported values of loss that were obtained using the old Marsh–Schulkin or Thorp attentuation formulas. Due to the depth dependence of a typical *p*H profile, this impact will depend on the configuration of the measurement in the water column. For example, previous values of surface loss that were obtained from propagation loss measurements in a surface duct (where the *p*H and, hence, the attenuation have the highest values) will be too high since the previously assumed value of attenuation was too low. Estimates are made of the possible impact on some of the most widely used formulas. [Work supported by NUSC.]

10:00

DD25. Spectral ratios of acoustic log in marine sediments. Jeffrey A. Meredith, C. H. Cheng (Earth Resources Laboratory, MIT, 42 Carleton Street, Cambridge, MA 02142), and Roy H. Wilkens (Hawaii Institute of Geophysics, Honolulu, HI 96822)

Diagnostic patterns have been noticed in spectral ratios of full waveform acoustic log data taken in slow marine sediments. The spectral ratio is computed with the full 2.56-ms time series after taking the Fourier transform and dividing the 3.05-m (10-ft) source–receiver offset transform by the 2.44-m (8 ft) one. The patterns show less attenuation at certain discrete frequencies in the spectra. The strongest observed peak in the spectral ratio was at slightly less than 7 kHz. Many of these patterns were duplicated by forward modeling consisting of calculating the branch cut integral of the compressional head wave at different source–receiver offsets and computing the spectral ratio. Preliminary analyses show that the presence and location of the peaks in the spectral ratio are controlled by the compressional wave velocity and the Poisson's ratio of the sediments, but the magnitudes of the peaks are controlled by their attenuation. The results were used to invert the full waveform log for shear wave velocity and compressional wave attenuation.

10:05

DD26. Transmission and reflection of acoustic waves through a composite slab using the finite element–eigenmode analysis. J.-H. Jeng, Vasundara V. Varadan, Vijay K. Varadan, and X.-Q. Bao (Department of Engineering Science and Mechanics, The Pennsylvania State University, University Park, PA 16802)

A numerical approach is presented for solving the interaction of acoustic waves with a composite slab that has periodically distributed substructures and is immersed in an infinite fluid medium. Reflection and transmission characteristics are studied for the case of normal plane-wave incidence. The mathematical model proposed combined the finite element method and eigenmode analysis, and can reduce the boundary value problem to a typical force vibration problem. Thus the damping of the composite slab can easily be introduced in the formulation. The results have been plotted as reflection and transmission coefficients versus frequency for different substructures and have been verified according to exact layer theory and two simple vibration models. Good agreement is observed. The presented analysis may be easily extended to the cases of oblique incidence and should be useful for future applications. [Work supported by the Research Center for the Engineering of Electronic and Acoustic Materials.]

WEDNESDAY MORNING, 15 NOVEMBER 1988

IAO NEEDLE/AKAKA FALLS ROOM, 9:00 A.M.

Meeting of Accredited Standards Committee S2 on Mechanical Shock and Vibration

to be held jointly with the

Technical Advisory Group (TAG) Meeting for ISO/TC 108 Mechanical Vibration and Shock

S. I. Hayek, Chairman S2

Applied Research Laboratory, The Pennsylvania State University, P.O. Box 30, State College, Pennsylvania 16801

D. F. Muster, Chairman, Technical Advisory Group for ISO/TC 108

4615 O'Meara Drive, Houston, Texas 77035

Standards Committee S2 on Mechanical Shock and Vibration. Working Group chairs will present reports of their recent progress on writing and processing various shock and vibration standards. There will be a report on the interface of S2 activities with those of ISO/TC 108 (the Technical Advisory Group for ISO/TC 108 consists of members of S2, S3, and other persons not necessarily members of those committees) including a report on the meeting of ISO/TC 108, held in Canton, People's Republic of China from 5–16 September 1988.

WEDNESDAY MORNING, 16 NOVEMBER 1988

KOHALA/KONA ROOM, 9:55 A.M. TO 12:00 NOON

Session EE. Architectural Acoustics V: Building Acoustic Studies and Material Measurement for Architectural Acoustics

Ronald A. Darby, Cochairman
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Contributed Papers

9:55

EE1. Acoustical design of Tokyo Geijutsu Bunka-Kaikan. Part 1. Study of room allocation for subway noise control and sound insulation. Satoru Ikeda, Keiji Oguchi, and Minoru Nagata (M. Nagata Acoustic Engineer & Associates Co., Ltd., 10 Shinano-machi, Shinjuku-ku, Tokyo, 160 Japan)

The Tokyo Geijutsu Bunka-Kaikan will be the second Tokyo Metropolitan Festival Hall (Bunka-Kaikan) and is scheduled to be opened in 1990. The building has a concert hall of 1887 seats, a theater of 850 seats, two small halls, and six rehearsal rooms. The basic problem with this building is the solid-borne noise from the subway running 3 m under the site. Therefore, the subway noise estimation was a major problem at the first stage of planning. Two types of hall allocation, horizontal and vertical, were studied, considering both subway noise control and sound insulation between auditoriums. The final plan was to place the concert hall on the top floor above the theater floor. The results of the investigation of the subway noise control and sound insulation structures will be presented.

10:07

EE2. An inquiry into noise intrusion in naturally ventilated tropical school buildings. Barbara Allen (School of Architecture, University of Hawaii at Manoa, Honolulu, HI 96822)

Many schools in tropical areas are naturally ventilated to save on costs of air conditioning. The inherent problem with this method of cooling is noise intrusion from adjacent areas, rendering the classrooms marginal teaching environments. Four naturally ventilated problem classroom buildings on the University of Hawaii campus are studied. The noise problems were identified as to source, location, SPL, and frequency. Reverberation times were measured for the classrooms at the various frequencies. Construction documents were used to verify the various building materials and that field transmission loss tests were conducted. Theoretical retrofit solutions were proposed for each classroom building. Solutions included site alterations, building fenestration additions, and interior space modifications. This study is the initial phase of the actual physical modeling that will take place next year in order to test our theoretical solutions.

10:19

EE3. Acoustical design of a multichannel broadcasting sound studio. Tohru Fukunishi, Toshio Wakatsuki, and Teruji Yamamoto (Department of Engineering Administration, NHK, 2-2-1, Jinnan, Shibuya-ku, Tokyo, 150-01 Japan)

NHK has constructed a multichannel sound studio for the purpose of efficient production of music programs. In order to provide musicians with well-defined sounds and to pick up well-separated sounds individually, the studio has a main area, three booths, and three alcoves suited for each use. The largest booth, with a floor area of nearly 92 m^2 and a height of 9.5 m, is for strings. It is acoustically separated from the main area by four sliding glass doors. Variable reverberation units installed on its walls can vary the reverberation time from 0.64 to 0.81 s at 500 Hz. The alcoves are designed so as to be usable as small separate booths, when necessary, by setting up two of the sliding glass doors, which can be housed in the inner walls. The floors of the electric instrument alcove and the drum booth are independently floated off the main floated floor. Furthermore, in this report, the details of the acoustical design, the inner sound insulation of each booth and alcove, the vibration proofing and noise control of the air-conditioning equipment, and the measurements of the acoustical characteristics are also mentioned.

10:31

EE4. A basic study of acoustical design for a four-channel sound control room. Toshio Wakatsuki, Tohru Fukunishi, and Teruji Yamamoto (Department of Engineering Administration, NHK, 2-2-1, Jinnan, Shibuya-ku, Tokyo, 150-01 Japan)

A number of experiments have been carried out in order to research how to design the interior of a four-channel sound control room for "high vision." In this four-channel sound reproduction form, the three main channels are reproduced from the three front loudspeakers, and the surrounding sound of the fourth channel is reproduced from the two rear ones. In these experiments, interiors of the test rooms were varied in accordance with five measuring conditions. However, room shapes were not varied. The experimental items were: (1) static transfer frequency response, (2) mean frequency spectrum of music sound, (3) questionnaire, and (4) subjective estimation. The results from the following three viewpoints were analyzed: (1) less coloration, (2) the width of the listening area, and (3) the amenities of the sound production room. It was found that when treating the front walls, ceiling, and floor with absorbent materials and providing the rear with a diffusive and reflective surface, the results were superior for all experimental items. It was also confirmed that this concept is applicable to ordinary two-channel sound control rooms. Experimental results in actual two-channel sound control rooms are also mentioned.

10:43

EE5. Evaluation of Japanese speech intelligibility for the sound system of an auditorium by STI. Keiji Oguchi and Minoru Nagata (M. Nagata Acoustic Engineer & Associates Co., Ltd., 10 Shinano-machi, Shinjuku-ku, Tokyo, 160 Japan)

Subjective hearing tests and STI measurements were carried out in a church and in six auditoriums. The relationship between the intelligibility and the directivity of the loudspeakers was studied for Japanese speech. In the church, the experiment was performed with four types of loudspeakers having different radiation directivity. At the point of $4*D_c$ (D_c is the critical distance) from the loudspeaker, STI was about 50%, and the intelligibility was found to be "fair." STI measurements were successively carried out in six auditoriums with clustered loudspeaker systems designed to cover the audience area within $4*D_c$. The results showed almost the same relationship between the STI and the ALcons given by T. Houtgast even for Japanese speech. Finally, the validity limits of STI as parameter for evaluating the performance of sound systems in auditoriums is discussed.

10:55–11:00

Break

11:00

EE6. Spatial Fourier transform method of measuring acoustic reflection coefficients at oblique incidence. Masayuki Tamura (Division of Systems Analysis and Planning, National Institute for Environmental Studies, Tsukuba, 305 Japan)

A new method using the spatial Fourier transforms has been developed to measure acoustic reflection coefficients at oblique incidence. The method involves the measurement of complex pressure fields in two parallel planes lying close to the surface of an absorbing material, and the decomposition of each of the complex pressure fields into plane-wave components by using two-dimensional spatial Fourier transforms. The incident and reflected plane-wave components on the surface of the absorbing material can be mathematically separated by using the plane-wave propagation theory. This separation leads to a determination of the acoustic reflection coefficients at arbitrary angles of incidence. Polyurethane foam was used as a sample material to check the validity of the method. A comparison between the results obtained by the present method and results from an existing free-field method showed good agreement except at very oblique angles of incidence.

11:12

EE7. Measuring acoustic impedance: The results of a worldwide round robin. James C. Haines (Manville Corporation, P. O. Box 5108, Denver, CO 80217) and Richard J. Peppin (Scantek, Inc., 51 Monroe Street, Suite 1606, Rockville, MD 20850)

ASTM Committee E-33 on Environmental Acoustics has published two different test methods for determining the normal incidence acoustic impedance and sound absorption coefficients of materials using an impedance tube. Method C-384 involves the use of the traditional standing-wave apparatus using pure tones. Method E-1050 involves the use of a two-microphone technique that provides improved measurement speed and increased detail in the performance versus frequency spectrum. Committee E-33 organized a round robin test program in order to generate ASTM required precision and bias statements for both test methods. Although not required by ASTM, the program also provided an excellent opportunity to compare the results obtained by the two measurement methods. Twenty-two laboratories expressed initial interest and fourteen laboratories from five countries throughout the world eventually participated in the program utilizing either one or both of the test methods. This paper discusses the organization of the program, the test results that were reported, and the conclusions that were drawn from a statistical analysis of the reported test results.

11:24

EE8. An estimation of absorption coefficients measured in a small reverberation chamber. Sojun Sato, Takeshi Fujimori, and Hajime Miura (Electrotechnical Laboratory, Tsukuba, 305 Japan)

This paper describes a simple and practical technique for measuring absorption coefficients of a material utilizing a small reverberation chamber. The chamber is used as a sound insulation box as well as a chamber for a reverberant sound field. The chamber does not have an ideal diffuse sound field but a sound field just for absorption measurements. A method based on the sound intensity technique using array microphones has been adopted to measure the absorption coefficients. Many loudspeakers are arranged in the chamber in order to improve the directivity distribution of the incident sound power near the surface of the absorbing material and to simulate a random incident sound field. Theoretical and experimental considerations regarding the effect of multiple sound sources in a small reverberation chamber and regarding the dependence of measured reverberant absorption coefficients on the distribution of sound power directivity are presented.

11:36

EE9. The three-microphone impedance tube. J. S. Bolton and M. Kompella (Ray W. Herrick Laboratories, School of Mechanical Engineering, Purdue University, West Lafayette, IN 47907)

The two-microphone impedance tube technique introduced by Seybert and Ross, and Chung and Blaser, is now widely used and has recently been standardized by the ASTM. This standard does not account for the effects of wall attenuation on measurement accuracy. Chu has recently indicated how two-microphone measurements may be adjusted to account for wall attenuation if the latter is known either from theory or independent measurement. In the work reported here, the effects of wall attenuation on the accuracy of the two-microphone technique are illustrated by use of computer simulations. The results show that the largest errors occur when the impedance of the test material is large; this is often the case at low frequencies for layers of porous materials, for example. It will be shown that measurements with three equally spaced microphones may be used simultaneously to determine both the complex amplitudes of the forward- and backward-going wave components and the complex wavenumber in the tube. From this information, the impedance of the test material may be determined accurately even in the presence of significant wall attenuation. In this way, the general approach of the two-microphone technique may be extended to allow measurement of higher impedance materials than has formerly been the case.

11:48

EE10. Random incidence sound transmission through foam-lined panels. N.-M. Shiau, J. S. Bolton, and D. A. Ufford (Ray W. Herrick Laboratories, School of Mechanical Engineering, Purdue University, West Lafayette, IN 47907)

Recent measurements of the sound transmission loss of aluminum panels lined with polyurethane noise control foam are reported. The measurements were conducted using an intensity probe to measure the sound power transmitted through panels mounted in the wall of a reverberation room in which a diffuse sound field was generated. Tests were conducted using four single-panel configurations: i.e., a single 1.27-mm aluminum panel and a 27-mm foam layer that was either bonded directly to one side or the other of the panel with glue or was separated from the panel by small spacers (thus creating a thin air gap between the panel and the foam). It was found that bonding the foam to the panel lowered the coincidence frequency from approximately 10 to 2 kHz; below this frequency, however, performance is improved relative to the unbonded treatment. Four double-panel configurations have also been tested: i.e., with the foam mounted directly to one or the other of the interior panel faces or separated from it by a small distance using spacers. Effects similar to those described above have been observed. It will also be shown that these effects are well predicted by a recent theory of two-dimensional wave propagation in elastic porous material [J. S. Bolton and N.-M. Shiau, AIAA Paper AIAA-87-2660 presented at the 11th Aeroacoustics Conference, San Jose, October 1987].

WEDNESDAY MORNING, 16 NOVEMBER 1988
KAUAI ROOM (WEST END), 10:00 A.M. TO 12:00 NOON

Session FF. Speech Communication VI: Production, Part B (Poster Session)

Keikichi Hirose, Cochairman
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Stefanie Shattuck-Hufnagel, Cochairman
Research Laboratory of Electronics
Massachusetts Institute of Technology
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Contributed Papers

Posters should be set up before 10:00 a.m. All posters will be displayed from 10:00 a.m. to 12:00 noon. To allow contributors the opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 10:00 to 11:00 a.m. and contributors of even-numbered papers will be at their posters from 11:00 a.m. to 12:00 noon.

FF1. Vowel durational characteristics during simultaneous communication. Robert L. Whitehead and Brenda H. Whitehead (National Technical Institute for the Deaf, 1 Lomb Drive, Rochester, NY 14623-0887)

The purpose of this investigation was to study vowel durations during speech produced simultaneously with sign language and fingerspelling and to determine whether final consonant influences on vowel durations, which occur naturally in speech, also occur in speech produced during simultaneous communication. Vowel durations were obtained for the vowels /ɑ/, /ɪ/, and /æ/, which were embedded in CVC words in which the final consonant was systematically varied. All CVCs met the criteria of having a common sign and of being phonetically and orthographically correlated. The CVCs were embedded in a carrier phrase and were uttered by ten normally hearing speakers fluent in sign language, under conditions of: speech, speech combined with sign language, and speech combined with signing of the carrier phrase and fingerspelling of the CVC. Results indicated that vowels produced with accompanying sign language and fingerspelling were significantly longer in duration than those produced during speech alone. Further, it was found that in both the signed and fingerspelled conditions, as in speech alone, vowels followed by a voiced consonant were longer in duration than vowels followed by their voiceless cognate.

FF2. A reexamination of the compensation effect in the mora hypothesis. Takashi Otake (Department of General Education, International Budo University, 841 Shinkan, Katsuura-shi, 299-52 Japan)

The objective of this paper is to reexamine the compensation effect of the mora hypothesis in mora timing. An early investigation of the mora hypothesis presented the compensation effect in Japanese as evidence in support of the mora hypothesis [Port *et al.*, Phonetica **37**, 235–252 (1980)] and another presented contrary evidence against it [M. Beckman, Phonetica **39**, 113–135 (1982)]. However, since Port *et al.*'s evidence was obtained in comparison with different test words produced by Japanese and Arabic speakers, their results need to be reexamined. In this paper, an experiment was conducted as to whether the compensation effect observed in Port *et al.*'s experiment will be relevant only to a mora-timed language by using the identical test words produced by Japanese and Arabic speakers. Contrary to Port *et al.*'s results, the results of this experiment indicated that the compensation effect was observed not only in Japanese but also in Arabic. These data indicate that the compensation effect may not be evidence for the mora hypothesis.

FF3. Durational effects of syllable structure and distinctive length in Japanese. William J. Poser (Department of Linguistics 2150, Stanford University, Stanford, CA 94305)

The results of a study of Japanese segment durations are reported in which segment identity was tightly controlled in order to reduce the effects of intrinsic segment duration, and distinctive vowel, while consonant length and syllable structure were varied. The corpus includes words with superheavy syllables, consisting of a long vowel followed by a geminate consonant, which have not previously been studied. One surprising result is the interaction observed when the durations of the components of superheavy syllables are compared to those of cognate bimoraic syllables containing a short vowel followed by a geminate consonant. The phonological lengthening of the vowel not only increases vowel duration as expected but also causes a substantial increase (25%–33% depending on the condition) in the duration of the already long consonant. In contrast, comparison of superheavy syllables with bimoraic syllables containing a long vowel followed by a nongeminate consonant showed no consistent effect of the phonological lengthening of the consonant on the duration of the vowel. This also shows that, while the mora count may determine the total duration of the word, the increment in duration due to the addition of a mora is not realized entirely in the segment(s) which that mora contains.

FF4. On sentential effects in the control of segmental duration in Japanese. Kazuya Takeda, Yoshinori Sagisaka, and Hisao Kuwabara (Speech Processing Department, ATR Interpreting Telephony Research Laboratories, Twin 21 MID Tower, 2-1-61 Shiromi, Higashi-ku, Osaka, 540 Japan)

Mainly, only word level factors, such as the temporal compensation between neighboring segments, have been used for duration control in the speech synthesis of Japanese. In this paper, the durational characteristics of Japanese are analyzed aiming at a higher level control, which reflects also sentential effects on segmental duration. First, the word level control was formulated as a linear additive model influenced by neighboring segments, location in a word, and word length. The parameters of the formula needed to integrate these features were obtained through factor analysis on a large-scale isolated word data base (5240 words). An experiment using 216 words showed that the formula was more accurate than the conventional method. Second, focusing on sentential effects could be accomplished by comparing the predicted with the measured durations of continuous speech. The prediction errors clarified the following characteristics of sentence level duration control. (1) Prepausal lengthening is greater than ordinary word final lengthening. (2) Shorter durations are found at sentence final position in declarative sentences.

FF5. Final lengthening and the prosodic hierarchy. Bruce Hayes (Department of Linguistics, UCLA, Los Angeles, CA 90024-1543)

Segments undergo phonetic lengthening both utterance-finally and in certain medial contexts, characterized informally as "word-final" or "phrase-final." No complete account exists of what these contexts are, or how they vary in the degree of lengthening induced. In phonology, prosodic hierarchy theory (Selkirk, 1980) posits that syntactic structure is reparsed into a hierarchy of nested levels of phonological phrasing. The theory is based on phonological rules that apply only when all segments involved occur within a single phrase of a particular size. The theory suggests a general account of final lengthening: The higher the level of a phrase on the prosodic hierarchy, the more final lengthening it receives. This hypothesis is compatible with the final lengthening rules proposed by Allen *et al.* (1987). The experiment described here used the method of Beckman and Edwards (forthcoming) to test the hypothesis further. Sentences were recorded in which the words *hate*, *incompetence*, and *hating competence* were divided across clitic group, phonological phrase, and intonational phrase boundaries. The prediction is that *hate* and *hating* should show increased final lengthening across the three contexts. Pilot results agree roughly with the predictions of the theory.

FF6. Durational rules from a speech data base. Rolf Carlson and Björn Granström (Department of Speech Communication and Music Acoustics, Royal Institute of Technology, Box 70014, S-10044 Stockholm, Sweden)

Rules for segmental duration have been studied in the context of a speech data base that is under development in our department. The data base search procedures include the same kind of context sensitive rules that are used in our speech synthesis project [Carlson and Granström, Phonetica **43**, 140–154 (1986)]. This affords the possibility of making a direct comparison to the data base durations when developing durational rules. Different kinds of speech material have been studied, including a novel and read sentences. Corresponding rules will be compared. Some different descriptive frameworks have been tried. A modified version of a rule structure suggested by Klatt has proven to be especially useful. [This work has been supported by grants from the Swedish National Board for Technical Development and the Swedish Telecom.]

FF7. Motor control mechanisms of stress production. Vincent L. Gracco and Patrick Haggard (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511)

For a variety of behaviors, emphasizing certain movements in a sequence can be used to establish perceptual saliency or vary communicative intent. The present investigation, part of a larger investigation of speech and nonspeech motor control mechanisms, was designed to evaluate modifications in movement sequences when subjects produced sentences under various emphatic stress conditions. Lip and jaw movements were compared using measures of displacement, velocity, duration, and relative timing of opening and closing. A secondary focus involved within-subjects comparison of the kinematic patterns of speech stress with those of stressed repetitive tapping. As in previous phrase-level speech investigations, local (stressed) and remote (nonstressed) timing effects were observed. In speech, all movements in the sequence were modified, not only the emphasized segment. The pattern of adjustments characterizing the local motor effects for speech was qualitatively similar to the motoric

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effects for stressed tapping. Emphasis of a particular component in a movement sequence may reflect a general motor strategy allowing flexibility while minimizing the number of basic motor processes. Specific kinematic adjustments will be discussed in relation to the underlying control mechanisms. [Work supported by NIH.]

FF8. Acoustic–phonetic correlates of stress shift. Stefanie Shattuck-Hufnagel (Research Laboratory of Electronics, Massachusetts Institute of Technology, 36-525, 77 Massachusetts Avenue, Cambridge, MA 02139)

Many speakers of American English exhibit "stress shift," in which the lexical prominence of a polysyllabic word seems to shift to an earlier syllable, as when "thirTEEN" becomes "THIRteen" in "THIRteen MEN," or "MissisSIPpi" becomes "MISsissippi MUD." Fundamental frequency and duration measurements suggest that this is a phrase-level phenomenon that arises from the confluence of two separate events: (a) The pitch accent occurs not on the target word, but on a different word of the phrase (here "MEN" and "MUD"); and (b) another phrasally based pitch rise occurs on the earlier syllable of the target word (here "THIR-" and "MIS-"). Three lines of evidence support this account: (1) The first rise can occur even earlier, i.e., on a preceding word, suggesting a domain larger than the individual lexical item; (2) under certain conditions of phrasal prosody, for the same word string, the pitch accent returns to the location of the lexically stressed syllable of the target word, showing that the "shift" does not result simply from the lexical stress patterns of adjacent words; and (3) the two types of pitch excursion can have different characteristics, indicating that the early rise and the pitch accent are separate phenomena. These observations are consistent with the hypothesis that pitch marking occurs on lexically stressed syllables when phrasal prosody requires it, but not otherwise.

FF9. Acoustic correlates of stress in four demarcative-stress languages. Haruko Kawasaki (Voice Processing Corporation, 1 Main Street, Cambridge, MA 02142) and Stefanie Shattuck-Hufnagel (Research Laboratory of Electronics, Massachusetts Institute of Technology, 36-525, Cambridge, MA 02139)

The acoustic correlates of stress in languages with "demarcative" stress (located at a fixed position relative to a word boundary) were investigated: two initial-stress languages (Finnish, Bengali) and two final (French, Hebrew). For each language, two to three speakers produced a frame sentence, then repeated it, replacing the three-syllable target word with /mamama/, /mimimi/, or /momomo/. Thirty reiterant versions from each speaker were analyzed for syllable duration, average intensity, intensity peak, intensity-duration integral, and location of $f0$ rise and $f0$ peak. It was hypothesized that initial-stress languages take advantage of higher $f0$ and intensity at utterance beginnings, while final-stress languages exploit the longer duration of utterance-final syllables. Results confirmed the association of longer duration with final stress but not of $f0$ and intensity with initial stress. Additionally, different syntactic/prosodic structures can change $f0$ patterns. For example, in French, one frame sentence caused $f0$ to rise in the last syllable, while another did not. This raises questions about the relative importance of different acoustic parameters for (a) the categorization of demarcative-stress languages, and (b) lexical versus phrasal prominence.

FF10. Stress and interstress intervals in reading. Gunnar Fant and Anita Kruckenberg (Department of Speech Communication and Music Acoustics, Royal Institute of Technology, Box 70014, S-100 44 Stockholm, Sweden)

Objective and subjective measurements of syllabic stress in reading of a Swedish novel have been performed. Our data bank contains material from 16 speakers. Measurements of durations and local pitch variations correlate well with the direct subjective estimate of stress. A comparison is made of subjects' introspective grading of stress from silent reading and their ability to discriminate individually produced stress patterns. The top-down influence is partial only. Statistics of interstress intervals show that relations between interval durations and the number of phonemes in an interval display speaker-specific characteristics. Moreover, the average of stress intervals not spanning across syntactic boundaries constitutes a rhythmical time constant of the order of 0.5 s in the planning of pauses, such that the sum of physical pause plus terminal lengthening tends towards an integral multiple, 1, 2, or 3, of the time constant. These data support and extend the model of W. A. Lea [*Trends in Speech Recognition* (Prentice-Hall, London, 1980)]. [Work supported by the Swedish Board for Technical Development and The Bank of Sweden Tercentenary Foundation.]

FF11. Temporal compensation in interaccent intervals and polysyllabic words in spoken French. Janet Fletcher (Centre for Speech Technology Research, University of Edinburgh, Edinburgh EH1 1HN, Scotland)

"Syllable-timed" descriptions of French assume a monotonic increase in interaccent interval and polysyllabic word duration with increasing syllable count, unlike "stress-timed" languages. Two sets of French data consisting of six excerpts of spontaneous speech and a corpus of nonsense word data were analyzed to test these claims. Acoustic measurements of nonprepausal syllable and interaccent interval duration in the first corpus revealed that the latter was positively correlated with the number of syllables contained. There was evidence of negative acceleration of increasing unit length, an effect not reported previously. Temporal compression was most evident between monosyllabic and disyllabic units, radically reduced between disyllabic and trisyllabic units, and completely missing beyond three syllables. The second corpus revealed similar patterns of compression in tokens of one to three syllables in length. These results are not unlike those reported by Port (1981) for American English although the "absolute compression levels" were smaller in these French data. This relates to the vowel-quality preserving nature of French in unaccented contexts as opposed to an "underlying rhythmic tendency" to equalize syllable durations.

FF12. Acoustic characteristics of affective prosody in young children. Christiane Baltaxe (Department of Psychiatry and Biobehavioral Sciences, Room 68-268, University of California, Los Angeles, CA 90024)

Acoustic characteristics of affective prosody in 30 children aged between 36 and 48 months, matched for SES and IQ, were measured in spontaneous and imitated utterances. Intonation patterns for neutral–normal, happy, sad, and angry affect were elicited and audiorecorded under controlled conditions. The responses were transcribed phonetically to account for speech variations relevant in the analysis. The recordings were analyzed acoustically for characteristics of fundamental frequency, intensity, duration, and their co-variation for each affective pattern using Oscillomink (Siemens) tracings and a PM pitch analyzer (voice identification). It was hypothesized that the production of affective prosody would parallel its perception. An earlier study (C. Baltaxe, *Proceedings of the Sixth International Phonology Meeting*, 1–7 July 1988, Krems, Austria) had shown that the same age group perceived the above affective patterns beyond chance. However, degree of success was linked to specific emotion, with a Polyanna effect toward happy emotion. The results of the present study showed that affective productions did not parallel perception, even in the imitation condition. Results are discussed with respect to articulatory control necessary for each pattern. [Work supported by Maternal Child Health.]

FF13. Prosodic components of speech in the expression of emotions. Yoshinori Kitahara and Yoh'ichi Tohkura (ATR Auditory and Visual Perception Research Laboratories, Twin 21 MID Tower, 2-1-61 Shiromi, Higashi-ku, Osaka, 540 Japan)

For the purpose of a natural and high-quality speech synthesis, the role of prosody in speech perception has been studied. Prosodic components, which contribute to the expression of emotions and their intensity, were clarified by analyzing emotional speech and by performing listening tests on synthetic speech. It has been confirmed that prosodic components, which are composed of pitch structure, temporal structure, and amplitude structure, contribute to the expression of emotions more than the spectral structure of speech. Listening test results also showed that the temporal structure was the most important for the expression of anger, while both amplitude structure and pitch structure were much more important for the intensity of anger. Pitch structure also played a significant role in the expression of joy and its intensity. These results suggest the possibility of converting a neutral utterance (i.e., one with no particular emotion) into utterances expressing various kinds of emotions. These results can also be applied to controlling the emotional characteristics of speech in synthesis by rule.

FF14. Pitch dynamism in bilingual females and males. Caroline G. Henton (Linguistics Program, University of California at Davis, Davis, CA 95616)

Pitch dynamism refers to how rapid or slow are changes in pitch range from high points to low points, and vice versa. Previous studies of pitch have focused largely on average pitch. Dynamism has been little explored instrumentally, and scarcely at all with regard to the variable speaker sex. Using recordings of female and male bilingual speakers, dynamism will be measured on the basis of first-differentiated fundamental frequency curves. These data will be used to explore whether there are intersex differences in pitch dynamism, and whether there is variation according to language spoken. Findings will be related to previously reported results for pitch ranges by females and males. Taking the combined results, it will be possible to respond empirically to stereotypes existing for more excitable or more variable speech for female speech, a response that has been urged by, e.g., Kramer (1977), McConnell-Ginet (1983), and Smith (1985).

FF15. A methodology for analyzing prosody. P. Price (SRI International, Menlo Park, CA 94025), M. Ostendorf (ECS Department, Boston University, 44 Cummington Street, Boston, MA 02215), S. Shattuck-Hufnagel (Research Laboratory of Electronics, Massachusetts Institute of Technology, 36-525, Cambridge, MA 02139), and N. Veilleux (ECS Department, Boston University, Boston, MA 02215)

Prosody consists of that information in speech which can be suprasegmental, i.e., operate over units larger than a single segment. Prosodic contours may span a word, phrase, or larger units. The goal of this study is to understand the mapping between discrete, abstract units (e.g., boundary tones, pitch accents) and their observed continuously varying acoustic correlates (e.g., duration and $F0$). Since prosody synthesis and analysis have traditionally been a challenging problem, the initial focus of this study has been restricted to the FM radio news broadcasting style of speech. This pilot study indicates that in this speech style, prosodic units appear to be more strongly and more regularly marked than in conversational styles. Time- and pitch-scale modification is employed using a sinusoidal model to change the duration and $F0$ contours of natural utterances. This technique allows synthesis of an utterance with a new prosodic contour, taking the input parameters from a linguistic model of prosody, or from another utterance with different phrasing. In this way, the perceptual effectiveness of various models in marking prosodic prominences and boundaries can be explored. The details of the method will be discussed, and a sample tape will be played.

FF16. Word structure and accentuation in Japanese—Is Japanese basic accent the antepenultimate type? Hirokazu Sato (NTT Human Interface Laboratories, 3-9-11 Midori-cho, Musashino, 180 Japan)

It has been reported that the accent nucleus of a Japanese word tends to be located on the antepenultimate mora, i.e., the third mora counted from the end of the word (henceforth, the antepenultimate type). This tendency of the accent pattern of Japanese has been confirmed through various statistical analyses [e.g., S. Hashimoto, Trans. Inst. Electron. Commun. Eng. Jpn. **56-D**, 654–661 (1973)]. In this paper, the relationship between word structure and accent pattern has been investigated using about 50 000 headwords in the *Japanese Accent Dictionary* to explain why the antepenultimate type appears to be the most frequent accent pattern in Japanese. The results show that this accent type mainly originates from compound accentuation in (full word + full word) and (full word + affix) concatenations, rather than being an intrinsic accent type of the Japanese language. Further study on the intrinsic accent of basic Japanese words of four morae or less has been conducted. The results will be reported according to the origin of the analyzed words, including original Japanese, Sino-Japanese, and loan words (from western languages).

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FF17. The influence of higher level linguistic information on production of duration and pitch patterns at syntactic boundaries. Cheryl M. Beach (Department of Psychology, University of Wisconsin—Madison, 1202 West Johnson, Madison, WI 53706)

Syntactic parsing strategies like "minimal attachment" have mainly been studied for written language. However, there is recent interest in whether parsing strategies are used for speech. An alternative view is that, because speech contains acoustic patterns that covary with structure, and these influence listeners' syntactic interpretations, acoustic patterns can indicate structure, and so strategies are unnecessary. A critical assumption of the latter view remains to be tested: Speakers reliably produce acoustic correlates of structure for minimal and nonminimal attachment sentences. The present study investigated this, comparing productions of minimal and nonminimal attachment sentences. Longer duration and more extreme pitch fall–rise were expected for a sentence containing a major syntactic boundary in the critical region (i.e., nonminimal attachment). Surprisingly, phrase-final lengthening did not appear; duration for the syntactic boundary condition was shorter. Additional experiments suggest why. When subjects assign sentence-focus-stress to material in the critical region of a no-boundary sentence, stress is represented by increased duration or pitch rise. Thus both syntactic and nonsyntactic linguistic information influence the form of the acoustic patterns produced, and determine whether the expected differences are observable across sentences. Implications for parsing models and for investigations of speech production and perception will be discussed.

Session GG. Engineering Acoustics IV: Electrical Acoustic Transducers

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Contributed Papers

2:00

GG1. The shortcomings of pure circuit models of transducers. Ilene J. Busch-Vishniac and Henry M. Paynter (Department of Mechanical Engineering, The University of Texas, Austin, TX 78712)

Circuit models of transducers have been used for decades and have proven to be useful tools for prediction of transducer behavior. In this approach, the transduction mechanism is typically modeled using ideal two-port transformers or gyrators that neither dissipate nor store energy. Energy loss and storage are taken into account using one-port elements connected to the transducing gyrator or transformer. An alternative approach to transducer modeling recognizes that the transduction mechanism itself generally is associated with energy storage or dissipation. In this formulation, the transduction mechanism is typically modeled using a two-port energic element (such as a two-port capacitance). A comparison of the two formulations for piezoelectric transducers shows that there are two fundamental shortcomings introduced in the circuit model approach that are not present in the energic multiport element approach: The transformer modulus must be made a function of the strain of the piezoelectric material, and it is not possible to discuss the piezoelectric material response without imposing a load on the transducer. It is concluded from this analysis that circuit models fail to represent accurately energy exchange in transducers.

2:12

GG2. A comparison of the absolute sensitivity of damped and undamped megahertz-range compressional ultrasonic transducers. Pamela D. Hanna,[a] William T. Yost, and John H. Cantrell (NASA-Langley Research Center, Hampton, VA 23665-5225)

An instrument and a technique for the absolute calibration of megahertz-range compressional wave transducers are reported. The methodology is based on the wave response characteristics of the previously developed submersible electrostatic acoustic transducer (ESAT). Measurements of absolute wave displacement amplitudes of the order of angstroms as a function of frequency are reported for both damped and undamped transducers immersed in water. Calculations of wave pressure, intensity, and corrections for diffraction based on these measurements are also presented. [a] Also at Analytical Services and Materials, 107 Research Drive, Hampton, VA 23665.

2:24

GG3. Composite slotted cylinder. T. A. Henriquez, Clementina R. Siders, and Robert E. Montgomery (Naval Research Laboratory, Underwater Sound Reference Detachment, P. O. Box 568337, Orlando, FL 32856-8337)

The composite slotted cylinder is a candidate for a low-frequency, high-powered, compact, underwater acoustic source that is being considered by the Naval Air Development Center. The geometry and boundary conditions of the slotted cylinder do not lend themselves to a complete closed-form solution. Therefore, a finite element model was employed to predict the behavior of a particular slotted cylinder configuration with and without radiation loading. Experimental modal analysis was used to confirm the assumptions used in the models of the unloaded transducer. The optimization of the acoustic performance was achieved using the finite element models developed. High-drive effects were also modeled. Comparisons are shown between predictive acoustic performance and the actual measurements. [Work supported by ONT.]

2:36

GG4. Microsurface acoustic wave resonator. G. Golan, G. Griffel, A. Seidman, and N. Croitoru (Department of Electronic Devices and Materials, Tel-Aviv University 69978, Israel)

Surface acoustic wave (SAW) resonators are the modern equivalents of the old bulk crystal resonators, used for high-Q frequency filtering, or very accurate frequency sources. A conventional resonator consists of a transmitting/receiving transducer, bounded from both sides by grid reflectors, creating a "Fabry–Perot" cavity structure. A major disadvantage of conventional resonators is their large dimension, which makes them inconvenient for integrated applications. In order to overcome this size limitation, new types of microresonators were designed and tested. These guided SAW resonators are able to guide and control the propagating component of the acoustic energy upon them, almost without losses. Based on theoretical calculations, two kinds of such devices were developed, namely, a corrugated waveguide filter and a microresonator structure. The main component in each of those devices is a metallic $\Delta V/V$ channel, which confines the propagating wave underneath and small, x-direction, transverse, short metallic strips in order to obtain periodic perturbations. Based on the "coupled mode theory," it was found that by using such periodic perturbations, a complete Fabry–Perot reflecting mirror for surface acoustic waves can be realized. The reflection coefficient of such a micrograting mirror is almost total and its performances are similar to the conventional resonators. [This work was supported by the Israeli Ministry of Communications.]

2:48

GG5. A new window function to synthesize a surface acoustic wave bandpass filter. Wu Kelin (Maanshan Transistor Factory, Anhui, People's Republic of China)

A new window function is proposed, which is written as $w(t) = [1 - th\, a(t/\tau)^2]$, where $a > 0$ and is a parameter to be selected according to the given response specification, and τ is the width of the window, namely, the length of the pulse response. The results of synthesizing SAWF by means of the new window function are reported. In synthesizing, one unapodized transducer and one length-weighted transducer are used. The transducers have a double electrode structure with each half of the double electrode individually weighted.

3:00

GG6. Loudspeaking telephone using echo cancellers and a voice-switching circuit. Hiroki Furukawa, Satoru Ibaraki, and Hiroyuki Naono (Development Research Laboratory, Matsushita Electric Industrial Company, Ltd., 1006 Kadoma, Osaka, 571 Japan)

A new loudspeaking telephone is proposed that makes duplex communication possible using an adaptive voice-switching circuit and two adaptive echo cancellers. An acoustic echo canceller with a delay capacity of 20 ms cancels the acoustic echo between a loudspeaker and a microphone. An electric echo canceller with a delay capacity of 8 ms cancels the electric echo caused by the impedance mismatch between the line and the balancing network. The characteristics of the acoustic echo path are subject to change by movement of the talker. To cope with such changes, two adaptive filters were installed in an acoustic echo canceller. At the start of talk, the voice-switching circuit inserts a large loss to suppress howling, because the echo cancellers do not replicate the impulse responses of the acoustic and electric echo paths at the beginning of the process stage. When the echo cancellers are replicating the impulse responses with the talker's voice, the inserted loss decreases to 3 dB. Duplex communication can be done in this way. An experimental circuit with three digital signal processors showed that duplex communication becomes possible after repeating voice transmission and reception a couple of times.

3:12

GG7. A study on howling suppression by automatic directivity control of a microphone array. Tadayori Makino and Takeshi Miyagawa (Corporate Engineering Division, Matsushita Communication Industrial Company, Ltd., 600 Saedo-cho, Midori-ku, Yokohama, 226 Japan)

Howling is one of the most annoying problems for public-address systems in a reverberant sound field. A new technique to suppress howling has been developed, in which the direction of the strongest coupling between the microphone and the loudspeaker is constantly detected to lower the sensitivity of the microphone in that particular direction. For detecting the direction of coupling, a reference signal, whose spectrum is modulated to spread over the frequency range of the public-address system, is combined with the acoustic input to be reproduced from the loudspeaker. This combined reference signal is eventually picked up by an array of microphone units. The direction of the coupling is detected from the difference in the arrival time of the reference signal between these microphone units. To lower the sensitivity of the microphone array for the particular direction, the directivity of the microphone array is controlled by individually delaying the outputs of each of the microphone units and then combining them as the output of the microphone array.

3:24

GG8. Optimization of a low-frequency transmitting array. Dominique Lalisse and Didier Boucher (GERDSM, DCAN Toulon, DCN, Le Brusc, 83140 Six-Fours-Les-Plages, France)

A low-frequency plane array is studied in water in a wide frequency band around the resonance frequency of the transducers. The array under study is made of eight length-expander vibrators (two columns of four transducers) with a circular radiating face in a rigid box of limited dimensions. The radiating impedance matrices are calculated by an integral equation method [C. Audoly, J. Acoust. Soc. Am. Suppl. 1 **83**, S20 (1988)] and projectors are modeled with a classical electromechanical equivalent circuit. Due to the effects of acoustic interactions, no terms in the matrices are found to be negligible. Mechanical and electrical constraints on the transducers are identified and computed. The array is studied under three conditions: identical voltage driving, identical headmass velocity distribution, and acoustic power optimization. The results confirm that acoustic interactions have important and drastic effects around the resonance. The study of acoustic power optimization makes it possible to discuss the opportunity of using velocity control and electromechanical feedback devices in low-frequency sonar projector arrays.

3:36

GG9. Numerical models used in design and analysis of volumetric transducer arrays. Scott A. Hudson (Ocean Systems Corporation (formerly Bendix Oceanics), 15825 Roxford Street, Sylmar, CA 91342-3597)

Analysis of transducer element and array performance has been aided by development of computer-based numerical models that include computation of mutual coupling effects. The farfield pressure distribution and array source level calculations necessitate accurate modeling of the individual transducers (in free field) as well as the varying radiation impedances due to the acoustic coupling between elements. Electromechanical equivalent circuits are employed to determine the complex drive voltages required to achieve velocity control within the array. Optimization methods that result in increased array source level and efficiency without forsaking velocity control have been investigated. This paper discusses numerical computations of farfield pressure, acoustic power, and mutual radiation impedances within volumetric projector arrays. Methods of optimization are also discussed. Predicted characteristics such as beam patterns and source level curves for various arrays of low-frequency flexural disk and flextensional transducers will be presented and compared to measurements.

3:48

GG10. A transducer for blood flow noise from intracranial vascular deformations. Kenji Kobayashi (Department of Electronic Engineering, Takushoku University, 815-1 Tatemachi, Hachioji, 193 Japan)

A blood flow noise transducer for use on the eyes has been designed to take the place of an ordinary accelerometer-type setting on the head skin. The blood flow noise is generally considered to be generated by the turbulent flow in an artery due to deformation of the vascular wall, such as an aneurysm. The present transducer is intended to detect the turbulent flow noise from the intracranial aneurysms clearly by setting the transducer on the eyes. A novel transducer has been constructed using PVDF film, which has a sensitivity of -63 dB (V/μbar) and a bandwidth of over 2.5 kHz. In clinical applications the transducer has shown fairly good results. Simulated experiments using the transducer with a cranial model suggest that the localization of the sound source can be predicted by data processing.

4:00–4:10
Break

4:10

GG11. Fiber-optic acoustic sensor using the Fabry–Perot interferometer. Katsunori Fujimura, Michio Matsumoto, Katsuji Hattori, and Hiroyuki Naono (Development Research Laboratory, Matsushita Electric Industrial Company, Ltd., 1006, Kadoma, Osaka, 571 Japan)

A highly sensitive and compact butt-coupled-type fiber-optic acoustic sensor has been developed based on the Fabry–Perot interferometer (FPI) for use in remote sensing or sensing in narrow spaces. The system consists of a sensor head, a controller, and an optical fiber transmission line. The sensor head is based on the FPI, and is formed from the high-reflectance-coated confronting surfaces of an optical fiber and a diaphragm. The sensitivity of the sensor head is related to the finesse of the FPI. As the finesse increases, so does the sensitivity. To ensure system stability and increase the dynamic range, the controller is composed of a DFB laser diode with a wavelength of 1300 nm, double optical isolators, an automatic temperature controller, and an automatic power controller.

4:22

GG12. A fiber-optic, interferometric, acceleration canceling, flexural disk hydrophone. David A. Brown, T. Hofler, and S. L. Garrett (Physics Department, Naval Postgraduate School, Monterey, CA 93943)

A fiber-optic hydrophone consisting of an air backed cylinder with clamped circular end plates bonded to four 8-m-long flat (spiral) wound single mode optical fibers will be described. For a given pressure differential between the inside and outside of the cylinder, the interior and exterior fiber coils on each 4.5-cm-diam, 1-mm-thick aluminum disk experience opposite strains. This "push–pull" design induces an optical path length difference in a Michelson interferometer. When subject to acceleration along the cylindrical axis, both disks are deflected in the same direction, producing a positive strain on the outer surface of one disk and an equal, but negative, strain on the outer surface of the other disk. Since the fiber coils on the exterior of the two disks comprise one arm of the interferometer (the interior coils comprise the second arm), the strains cancel. The calculated strain for this four-coil sensor [A. E. H. Love, *A Treatise on the Mathematical Theory of Elasticity* (Dover, New York, 1944), 4th ed., Sec. 309] and optically measured strain are in good agreement and yield a sensitivity of 0.5 rad/Pa below the plate resonance frequency of 4.5 kHz.

4:34

GG13. Theoretical analysis of a class V flextensional fiber-optic interferometric hydrophone. S. L. Garrett (Physics Department, Naval Postgraduate School, Monterey, CA 93943) and D. A. Danielson (Mathematics Department, Naval Postgraduate School, Monterey, CA 93943)

When an oblate spheroidal shell having an aspect ratio $a/b > \sqrt{(2-\nu)}$, where ν = Poisson's ratio, is subject to hydrostatic compression, the major (a) and minor (b) axes experience strain of opposite sign [R. A. Clark and E. Reissner, J. Mech. Phys. Solids **6**, 63 (1957)]. If two optical fibers, which comprise the arms of an interferometer, are wound around the equatorial and meridional circumferences of the spheroid, pressure changes induce a differential optical phase shift in the interferometer. Calculations of circumferential strain and of the optical pressure sensitivity and buckling pressure of such a sensor will be presented for thin shells ($t/b \ll b^3/a^3$, where t is the shell thickness). Sample designs based on these calculations will be compared to other fiber-optic sensors of similar dimensions and materials.

4:46

GG14. Piezoelectric transducer analysis using the transfer matrix method and a new versatile computer code. Pascal Tierce, Koffi Anifrani (SINAPTEC, 41 Boulevard Vauban, 59800 Lille Cedex, France), Jean Noel Decarpigny, B. Hamonic (Acoustics Laboratory, ISEN, 41 Boulevard Vauban, 59046 Lille Cedex, France), and Didier Boucher (GERDSM, Le Brusc, 83140 Six-Fours-Les-Plages, France)

The analysis of many simple magnetostrictive or piezoelectric transducers used in underwater acoustics, macrosonics, acoustical imaging, or nondestructive testing can be performed under a classical plane-wave assumption with the help of the transfer matrix approach [G. E. Martin, J. Acoust. Soc. Am. **35**, 510 (1963); **36**, 136 (1964); **36**, 1496 (1964); D. F. McCammon and W. Thompson, Jr., J. Acoust. Soc. Am. **68**, 754 (1980)]. Described here is a new versatile computer code, ASTRE, which relies on this approach and which can be run efficiently and interactively on any mainframe, workstation, or PC. In the mathematical model, the transducer is split into homogeneous elastic magnetostrictive or piezoelectric blocks that are cylinders, parallelepipeds, or conical frustra. These blocks are assembled, either in series or in parallel, without any restriction related to the connectivity. Internal losses are included and various terminal radiation or mechanical impedances are available. The harmonic analysis provides the displacement, strain, and stress fields as well as the electrical impedances, the source level, or the receiving sensitivity, depending upon the use. External electronic circuits can be added at the electrical ports, particularly useful in the receiving mode. Several different analyses are described in detail to demonstrate the capabilities of the code.

4:58

GG15. Acoustic wave field analysis of ultrasonic transducers. Shung H. Sung (Engineering Mechanics Department, General Motors Research Laboratories, Warren, MI 48090-9057)

A numerical method was developed for predicting the radiated acoustic field from arbitrarily shaped ultrasonic transducers. The numerical method was first verified for a spherical transducer, and it was found that the numerical results agreed very well with the exact analytical results. The numerical method was then applied to other transducer shapes where exact analytical results are not available: toroidal, elliptical, conical, and parabolic transducers. Among these transducers, the elliptical and the parabolic ones have never been studied in regard to their ultrasonic radiation characteristics. As expected, an ultrasonic transducer's geometrical shape affects the radiated acoustic field and acoustic beam pattern, and thus characterizes its specific application. For both the focusing- and non-focusing-type transducers, the ultrasonic characteristics of focal length, depth of field/depth of penetration, beam radius, sidelobe level, and its radial location were calculated and are discussed.

5:10

GG16. Piezoelectric thick copolymer for acoustic transducers. F. Bauer (Institut Franco Allemand, 12 Rue de l'Industries, 68301 Saint Louis Cedex, France), M. Boisrayon, M. Richard (GERDSM, Le Brusc, 83140 Six-Fours-Les-Plages, France), and C. Massot (Science Industries, 22 Avenue Liberation, 77130 Montereau, France)

Vinylidene fluoride/trifluoroethylene (VF2/VF3) copolymers with more than 50% of VF2 mole content are of ferroelectric nature. They can be formed into plates from 1 to 500 μ thick from the molten state. Studies conducted at ISL show that such copolymers can be poled to a high level of remnent polarization $P_r \geqslant 7\ \mu C/cm^2$. They exhibit a high piezoelectric activity, which is suitable for applications in acoustic detection, hydrophones, pressure measurements, etc. Piezoelectric properties are presented and discussed. In order to increase the capabilities and the performance of such copolymers VF2/VF3 with 70% mole content for acoustic transducers, copolymer plates 3 mm thick have been realized and stabilized. The poling process of an active area of 25 cm^2 was done by following the ISL process. Piezoelectric copolymer plates were obtained successfully. The characterization and piezoelectric properties d_{33}, kT, dh of such new active piezoelectric elements are presented. The results are discussed.

5:22

GG17. Critical piezoelectricity in percolation. Didier Sornette (Physique de la Matière Condensée, CNRS UA190, Faculté des Sciences, Parc Valrose, 06034 Nice Cedex, France), Michel Lagier (Thomson-Sintra, DASM, zone des Bouillides, Sophia-Antipolis, 06561 Valbonne, France), and Stéphane Roux (Laboratoire d'Hydrodynamique et Mécanique physique, CNRS UA 857, Ecole Supérieure de Physique et Chimie Industrielle, 10 Rue Vauquelin, 75231 Paris Cedex 05, France)

The problem of the critical behavior of piezoelectricity in percolating media is introduced for the first time. Its relevance stems from the fact that piezoelectricity associates mechanical and electrical properties whose separated critical behavior have been extensively discussed in the literature. It should also yield useful information on the deformation modes of the structures. This problem is suggested by the recent development of low-density, compliant flexible piezoelectric ceramics used as acoustic transducers. The physical properties of these systems are just beginning to be explored experimentally, and their modelization is an open problem. The limiting case of strong heterogeneousness, where percolation ideas may apply and lead to simple and universal predictions, is addressed. The simplicity of the analysis comes from the clear separation of three scales, the microscopic grain size R, the percolation mesh size ξ, which diverges as the percolation threshold is approached, and the macroscopic size L of the system ($R \ll \xi \ll L$). The main result is that the direct piezoelectric effect exhibits a critical divergence as more and more tenuous compliant porous ceramics are considered.

5:34

GG18. Transducer nonlinearity theory. Y. F. Chou (Shanghai Acoustics Laboratory, Academia, Sinica, No. 456 Xiao Mu Qiao Road, Shanghai, People's Republic of China)

One normally regards a transducer as a linear device. However, as the working frequency of nonlinear parametric arrays becomes lower and the parametric arrays nearfield theory becomes popular, the nonlinear effect of a transducer cannot be tolerated and seriously affects the outcome of the parametric measurement. The present theory regards the transducer linearity and nonlinearity in the following ways: First, the so-called "linearity" is the fact that the transducer will induce some quantity of electricity when there is sound pressure applied on the surface of the transducer, and the transducer can be regarded as a static capacitor. As for the nonlinear process of the transducer, the transducer not only induces some quantity of electricity but also vibrates in the direction of the sound wave, so the transducer is regarded not as a static capacitor but as a time-varied capacitor controlled parametrically by the incident sound pressure. The following conclusions are reached: (1) The transducer nonlinearity is independent of the polarization voltage; (2) the transducer nonlinearity is not only related to the difference frequency but also to the primary frequency; (3) it is a function of the first-order sensitivity of the primary frequency; and (4) the transducer nonlinearity also depends on the transducer characteristics and its shape.

5:46

GG19. An estimation of the characteristics of an ultrasonic transducer by means of waveform analysis of its transient response. Daitaro Okuyama and Yoshizo Mori (Mining College, Akita University, Akita, 010 Japan)

A transient current is generated when a steplike voltage is applied to a piezoceramic transducer. It is shown that the frequency spectrum and the damped capacity obtained by an FFT analysis of the waveform have a close similarity to the frequency characteristics of the admittance and the damped capacity as measured by an ordinary method. To generate the steplike voltage with a 0.01-μs rise time, a mercury relay and a power source, whose internal impedance is less than 1 Ω, are used. The large capacity (about 500 μF) is connected parallel to the output terminal of the dc power source to obtain such a low impedance. The damped capacity is calculated from the data length, the value of the applied dc voltage, and the dc component of the frequency spectrum. The resonant characteristics of the transducer can be obtained from the results of FFT analysis on the transient current waveform. Results for various types of transducers, such as piezoceramic disks, cylinders, and honeycombs, etc., obtained by the present method show good agreement with those obtained under the ordinary techniques.

WEDNESDAY AFTERNOON, 16 NOVEMBER 1988 KOHALA/KONA ROOM, 2:00 TO 6:00 P.M.

Session HH. Musical Acoustics II: Digital Signal Processing in Music

Isao Nakamura, Cochairman
University of Electrocommunications
1-5-1 Chofugaoka
Chofu, 182 Japan

James W. Beauchamp, Cochairman
School of Music
University of Illinois
Urbana-Champaign
Urbana, Illinois 61801

Chairman's Introduction—2:00

Invited Papers

2:05

HH1. Partial synchrony in musical sounds: Some recent results using time-variant spectral analysis. James W. Beauchamp and Robert C. Maher (School of Music and Department of Electrical and Computer Engineering, University of Illinois at Urbana-Champaign, Urbana, IL 61801)

Time-variant techniques have been used to analyze quasiharmonic musical signals in terms of

$$s(t) = \sum_{k=1}^{n} c_k(t)\cos\left(2\pi\int f_k(t)dt + \theta_k\right).$$

One type of synchronous behavior occurs when each harmonic amplitude obeys a definite monotonic relationship $c_k = F_k(c_1)$. This appears to be approximately true for nonvibrato reed-instrument tones, with the notable exception of their attack epochs. On the other hand, the functions relating the harmonic amplitudes of flute tones exhibit considerable hysteresis. Another type of synchrony has to do with harmonicity and whether the frequencies of the partials are harmonically locked together so that $f_k(t) = kf_1(t)$. Partials of vocal tones with vibrato appear to obey this relationship, but those of oboe tones do not. Examples of the analysis and synthesis of sounds with and without synchronous behavior will be given.

2:30

HH2. Digital signal processing aspects of digital musical instruments. Jun-ichi Fujimori and Hirokazu Kato (Insoft System Laboratories, Center for Musical Instrument and Software Development, Yamaha Corporation, 10-1 Nakazawa-cho, Hamamatsu, 430 Japan)

Digital signal processing techniques are being widely used in electronic musical instruments and their accessories. Today's digital musical instruments provide a greater variety of sounds and a more precise control than ever before. However, these instruments lack intuitive, direct controls making them difficult for most musicians to use. Two possible approaches are being considered to reduce their complexity. One is to utilize a powerful MPU for controlling the parameters of the digital signal processing, and the other is to find models of digital signal processing that are specific to music. In this paper, several techniques are surveyed, especially as applied to musical activities. In particular, some sound alteration methods are examined, leading toward a solution to the problem of the musician–machine interface.

2:55

HH3. Analysis and synthesis of sounds using waveform and source–filter models. Xavier Rodet (IRCAM 31 Rue Saint Merri, 75004 Paris, France)

Sound analysis and synthesis for music requires accurate analysis and high-quality synthesis of sounds. Some of the methods that were developed at the Institut de Recherches et de Communication Acoustique/Musique (IRCAM) for contemporary music research and production are described. The following analysis techniques have been implemented and used: FFT and phase-vocoder, wavelets, linear prediction algorithms (Levinson, global covariance, Burg, lattice covariance, lattice recursive), prony analysis, pitch detection, temporal envelope tracking (power tracking), detection of harmonic/noisy components, modal analysis (resonance modeling), formant-waveforms (FOF) coding, and formant trajectory coding. The phase-vocoder is used extensively for analysis and processing (filtering, cross-synthesis, time warping, etc.) of speech and music. Different LPC analysis algorithms have been studied and are used for precise estimation of spectral envelopes. These envelopes can be coded into (as well as recalculated from) formant parameters without loss of precision. Sounds and LPC residuals are coded into a *harmonic/noise* index for each of the frequency bands centered around harmonic partials of the signal, thus improving the quality of synthetic sounds. For sound processing, the following algorithms have been used: cascade filtering (LPC-vocoder), parallel filtering (in the program CHANT), frequency domain filtering (phase-vocoder), time-warping (phase-vocoder and LPC-vocoder), time-varying and arbitrary sample-rate conversion. Several synthesis methods have been developed, including LPC-derived synthesis filter, formant synthesis, formant-waveform (CHANT), parallel filtering, phase-vocoder, and additive synthesis. Synthesis-by-rule uses sophisticated object-oriented languages (PREFORM) for calculation, manipulation, and real-time control of synthesis parameters. For these studies, a real-time workstation is under development, based on SUN-3, Mercury array processor, and Sony PCM AD/DA.

3:20

HH4. String simulation by means of digital waveguides. Julius O. Smith (NeXT/CCRMA, 3475 Deer Creek Road, Palo Alto, CA 94304)

Research in digital sound synthesis and reverberation at CCRMA has led to a new approach, called "waveguide synthesis," in which acoustic traveling waves are explicitly simulated using digital delay lines. So far, the best success has been obtained for one-dimensional wave propagation, such as in woodwinds and strings. The ideal string is simulated using two delay lines, where one delay line carries the left-going traveling-wave component, and the other carries the right-going wave. Realistic strings are obtained by perturbing this lossless prototype using low-order digital filtering in cascade with the delay line. Because strings in musical instruments are close to lossless, it is highly efficient to simulate them as a weak perturbation of a lossless system. This paper reviews the fundamentals of string simulation using digital waveguides.

3:45

HH5. Rendering piano music via a performance model. Tomoyasu Taguti (Department of Applied Mathematics, Konan University, Kobe, 658 Japan)

A note-list approach for automated music performance on the piano is dealt with. First, a music language that provides the means for a high-level, quantitative description of the musical gesture is presented. It is characterized by (1) the organization of notes into a hierarchical data structure, and (2) the parametrization of performance (or interpretive) practice into terms of dynamics, agogics, articulation, shift, and damper motion. In this language, the notes of a given score can be structured in different ways according to one's musical interpretation, and the parametrized performance practice in each category can be given to any substructures. In this respect, an instance of note-list structure is viewed as a "model of performance" where the parametrized performance practices stand for the parameters of the model. Second, certain empirical rules for such "gestural parameters" are discussed in relation to the data analysis of performed music by professional pianists. [Work supported by the Grant in Aid for Scientific Research, Ministry of Education.]

4:10–4:20
Break

Contributed Papers

4:20

HH6. Digital sound researcher's workbench. Tatsuya Aoyagi (Department of Information Engineering, The University of Electrocommunications, 1-5-1 Chofugaoka, Chofu, 182 Japan) and Keiji Hirata (NTT Software Laboratories, 3-9-11 Midoricho, Musashino, 180 Japan)

The purpose of the *ICOTone* project is to develop a workstation for computer music. This paper describes the digital signal-processing aspect of the *ICOTone* system. There are many well-known techniques established for digital sound synthesis/analysis. However, there are few computer systems in which all of the techniques can be used simultaneously and easily. In the area of statistics, the S system is a good example; this was developed at the AT&T Bell Laboratories and is an interactive environment for data analysis and graphics. In the S system, a user can grasp what statistical data really mean very quickly because of the rich command library and the preferable man–machine interaction. The goal here is, roughly speaking, to build a digital sound system that employs the design principle of the S system. Such a system will be a common workbench for digital sound researchers. The system configurations will be shown briefly. At present, the *ICOTone* system includes SUN workstations, digital audio tape recorders (DAT), and compact disk players (CD). The software is implemented on the SUN workstations, using the X-window facilities.

4:32

HH7. Musicians' and nonmusicians' sensitivity to differences in music performance. J. Sundberg, A. Friberg, and L. Fryden (Department of Speech Communication and Music Acoustics, KTH, Box 70014, S-100 44, Stockholm, Sweden)

A set of ordered, context-dependent rules for the automatic transformation of a music score to the corresponding musical performance has been developed, using an analysis-by-synthesis method [J. Sundberg in *Generative Processes in Music*, edited by J. Sloboda (Clarendon, Oxford, 1987), pp. 52–69]. The rules are implemented in the LeLisp language on a Macintosh microcomputer that controls a synthesizer via a MIDI interface. The rules manipulate sound level, fundamental frequency, vibrato extent, and duration of the tones, though only by a barely audible quantity, so as to avoid exaggerated effects. However, for any rule, the quantity can be varied. The present experiment was carried out in order to find out if the sensitivity to these effects differed between musicians and nonmusicians. Pairs of performances of the same examples were presented in different series, one for each rule. Between the pairs in a series, the performance differences were varied within wide limits and, in the first pair in each series, the difference was great, so as to catch the subject's attention. Subjects were asked to decide whether the two performances were identical. The results showed that musicians had a clearly greater sensitivity. The pedagogical implications of this finding will be discussed.

4:44

HH8. Calculation of a musical Fourier transform. Judith C. Brown (Media Lab, MIT, Cambridge, MA 02138 and Physics Department, Wellesley College, Wellesley, MA 02181)

The discrete Fourier transform is inappropriate for the description of musical sounds in that it calculates frequency components for frequencies separated by the same absolute frequency difference and with the same resolution. Frequencies of musical interest, i.e., the notes of the equal-tempered scale, are geometrically spaced and one would like a resolution of at least a semitone (6% of the center frequency) independent of center frequency. Using a window size varying linearly with frequency over a low and a high region of musical relevance, and with center frequencies logarithmically spaced, a constant Q Fourier transform has been calculated. Applications to sounds generated by a variety of musical instruments will be presented. Since the frequency components of any sound with harmonic partials have the same spacing when plotted against $\log f$, this calculation is promising as a "pattern recognition" method of pitch tracking. Results on the violin and the flute will be presented. [Work supported by a Brachman Hoffman fellowship from Wellesley College.]

4:56

HH9. A study of the transient sounds of the shakuhachi based on ARMA modeling with residual excitation. Michiko Toyama (1-29-8 Hanegi, Setagaya-ku, Tokyo, 156 Japan)

Although the conventional method of linear predictive analysis is quite widely used in speech analysis and synthesis, the model used in such analysis is rather rigid and lacks flexibility in representing and reproducing the wide range of sound quality found in music. In this study, an attempt was made to use the method of ARMA modeling and excitation by the prediction residual signal, rather than by white noise and periodic pulses, especially for the analysis and synthesis of the sounds of the shakuhachi, a type of Japanese flute. Two taped tones, a weak D (293 Hz, apertures closed) and a strong G with "kazaiki" or breath noise (392 Hz, two apertures open) were used to study the characteristics of the noise section, the transient, and the steady state. The experimental work is focused on the pattern of the residual signal and on the possibility of its use as the excitation signal, replacing the white noise and pulse, in order to produce a variety of musical sounds. As for the modeling, the results indicate that more experimental studies are required.

5:08

HH10. Discrepancies between published piano scores and composers' recorded performances—Data obtained by a computerized FFT transcribing device. Mizue Katoh (Tokyo Disneyland, Chiba, 272-01 Japan)

The differences between published piano scores and recordings of the same music performed by its composers are presented. In order to demonstrate objectively such discrepancies, recorded performances of Debussy, Rachmaninoff, and Gershwin were transcribed by a personal computer (NEC-9801) using the fast Fourier transform. In order to verify the findings, the data obtained from Debussy's "La plus que lente" was reproduced on an electronic piano via a computer. The most outstanding differences ascertained were as follows. (1) Notes not written were played. (2) Notes written were omitted. (3) Notes to be played simultaneously were played separately. (4) Tempo and rhythm differed from those indicated on the score. As well as an analysis of the performances of the composers themselves, the FFT transcribing technique can be applied to the analysis of jazz and other music forms of irregular tempo, such as various types of non-Western music. In addition, the results of the FFT analysis can contribute to the fields of musical education, musicology, and so on.

5:20–6:00

Musical Performance. Two invited musical performances will follow the session. The first will be on the Kyoto, a Japanese strirg instrument, by Masateru Ando of the Tokyo University of Arts and Music. This will be followed by a demonstration of historical tunings and temperaments from a performance perspective by E. L. Kottick of the Iowa University School of Music.

WEDNESDAY AFTERNOON, 16 NOVEMBER 1988 OAHU/WAIALUA ROOM, 2:00 TO 4:23 P.M.

Session II. Noise II: Hearing Conservation: Evaluating and Preventing Occupational and Nonoccupational Noise-Induced Hearing Loss

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Hiroyuki Zusho, Cochairman
Department of Oto-Rhino-Laryngology
Kanto Rosai Hospital
Kawasaki, 211 Japan

Chairman's Introduction—2:00

Invited Papers

2:05

II1. Conservation and compensation for noise-induced hearing loss in Japan. Hiroyuki Zusho (Department of Oto-Rhino-Laryngology, Kanto Rosai Hospital, Kawasaki, 211 Japan)

In Japan there are workplaces with a high level of noise exceeding 100 dB(A) in various industries, such as shipbuilding, metal mining, metal manufacturing, and car manufacturing. In the past several years, the number of new beneficiaries of workers' compensation insurance for noise-induced hearing loss has ranged from 400 to 800 yearly. According to Japanese laws concerning conservation of hearing in workplaces, workplaces producing noise are designated as such. These places are obligated to measure the noise level periodically and also to carry out the control of the work environment, hearing conservation, and education on labor hygiene. Recently, a new guideline for noise control was published and hearing conservation in these designated workplaces will be carried out in accordance with this guideline in the future. As for workers' compensation insurance, various types of benefits are paid upon the worker's retirement in accordance with the extent of the hearing loss suffered by a worker at a workplace with a level of noise exceeding 85 dB(A) over a long period of time.

2:25

II2. Regulatory control over hearing conservation programs—The British Columbia experience. Margaret E. Roberts (Hearing Conservation Section, Workers' Compensation Board of British Columbia, 6951 Westminster Highway, Richmond, British Columbia V7C 1C6, Canada)

Regulations requiring hearing conservation measures have been in effect in British Columbia, Canada, for 10 years. All programs are under the supervision of the Workers' Compensation Board staff who ensure that technician training, equipment standards, and follow-up procedures are consistent throughout the province. All data are submitted to the Workers' Compensation Board main computer system. Reports for scheduling and summarizing program results by occupation, firm, and industry are returned to each company. Experience over the past 10 years has shown that cooperation between workers and management, plus coordination and control by a central regulatory body, have provided an atmosphere for the development of a stable and accepted hearing conservation program for British Columbia's industrial workers.

2:45

II3. Age variation in the upper limit of hearing. Shintaro Takeda, Ikuharu Morioka, Yoshiaki Yoshida, and Kenji Matsumoto (Department of Hygiene, Wakayama Medical College, 27 Kyubancho, Wakayama, 640 Japan)

The equipment for measuring the upper limit of hearing was produced to estimate hearing changes with age. The upper limits of hearing were individually measured with 4293 audiometrically, otologically, and neurologically normal ears ranging in age from 5–85 years. The results were as follows. In each age group of both sexes, the upper limits of hearing showed an approximately normal distribution if a logarithmic scale was used for the age axis. The median values of the distributions shifted to a lower frequency, and their deviation became larger, with increasing age. It is of much interest that from early childhood the deterioration of the upper limit of hearing is already in progress. Standard upper limit age curves were established by calculating 10, 25, 50, 75, and 90 percentiles for each age group.

3:05

II4. Procedures evaluated by the ANSI S12.12 Working Group for determining the effectiveness of hearing conservation programs (HCPs). Larry H. Royster (Department of Mechanical Engineering, North Carolina State University, Raleigh, NC 27695-7910) and Julia Doswell Royster (Environmental Noise Consultants, Inc., Cary, NC 27512-0144)

The ANSI S12.12 Working Group has evaluated several procedures for conducting audiometric database analysis (ADBA), including threshold shift criteria and indices of threshold measurement variability. Findings from applying these procedures to a large audiometric database created by the committee have been used to establish trial guideline ranges for judging HCP effectiveness. The S12.12 Working Group will publish its recommendations for trial utilization and evaluation. This paper presents the ADBA procedures evaluated by S12.12 and the trial guideline ranges under consideration. A recommended format for applying ADBA to evaluate the effectiveness of HCPs includes: (1) restricting the database to a known controlled population, (2) checking the database for non-noise-related contaminations such as calibration errors, (3) applying the procedures to the total database to judge overall program acceptability, and (4) using the procedures to compare the relative effectiveness of the HCP in protecting subpopulations such as groups in different production units or groups using different hearing protectors.

3:25–3:35
Break

Contributed Papers

3:35

II5. Development of a comprehensive hearing conservation computer system. Christine Dixon-Ernst, H. D. Belk, Mark Glover, Emma Pessolano King, and Nancy B. Sussman (Corporate Medical Department, Alcoa, 1501 Alcoa Building, Pittsburgh, PA 15219)

Development of a comprehensive hearing conservation computer system on a personal computer was done to facilitate administration of Alcoa's hearing conservation program. This system is based on a personal computer link to both a microprocessor audiometer and mainframe computer. The personal computer is linked to a microprocessor audiometer to perform automated audiometric tests. Test results are immediately analyzed using an algorithm that determines OSHA standard threshold shifts as well as shifts using criteria developed by Alcoa. Results and recommendations are generated in a letter following the test. Reports are made to both medical and IH personnel to assist in administration of the hearing conservation program. Test results are batched monthly via dedicated lines to a mainframe computer at corporate headquarters while complete demographic updates on plant individuals are batched down to the PC. This link to the mainframe provides accurate demographics, and relieves the nurse from data entry and maintains a central file of audiometric results for company-wide data analysis.

3:47

II6. Comparison of a nonlinear versus a linear earmuff for prevention of temporary threshold shift (TTS) from gunfire and user acceptability for speech understanding. Larry H. Royster (Department of Mechanical Engineering, North Carolina State University, Raleigh, NC 27695-7910), Julia Doswell Royster (Environmental Noise Consultants, Inc., Cary, NC 27512-0144), Elliott H. Berger (E-A-R Division, Cabot Corporation, Indianapolis, IN 46268-0898), W. Dixon Ward (University of Minnesota, Minneapolis, MN 55414), and Richard A. Nuss (E-A-R Division, Cabot Corporation, Indianapolis, IN 46268-0898)

Students in firearms classes wore either their regular earmuff or a prototype nonlinear earmuff during firing practice. Audiograms were performed before and after firing to detect TTS. Dosimeters were used to measure L_{eq} noise exposures and peak SPLs for two ammunition types (mean peak = 159 dB, standard deviation = 4.1 dB). Individuals shot an average of 110 rounds, with exposure affected by all 10–12 class members. Audiometric results showed no mean TTS for wearers of the linear earmuff, and a small but statistically significant mean TTS at 4, 6, and 8 kHz for wearers of the nonlinear muff (means <3 dB, standard deviations <5.5 dB). Although the nonlinear muff was not as protective against higher peak SPLs, students' questionnaire responses showed that they preferred the nonlinear muff for understanding speech, especially students with preexisting hearing loss.

3:59

II7. Workplace evaluation of earmuff-type hearing protectors. George Durkt and Dennis A. Giardino (Mine Safety and Health Administration, 4800 Forbes Avenue, Pittsburgh, PA 15213)

The measurement of attenuation values for hearing protectors as used by personnel at the work site give a more realistic estimation of the protection afforded the worker. Head movement, sweating, chewing, and other physical factors may affect the integrity of the hearing protector resulting in different attenuation values than those obtained from the standard ANSI test method. A miniature, two-channel, frequency-modulated, telemetry system has been developed to measure attenuation values for earmuff-type protectors. Data from small microphones located inside and outside the hearing protector are transmitted to a remote receiver (up to 200 ft) and tape recorder for later analysis. The wireless link between the worker and observer allows the gathering of data while leaving the worker unencumbered to perform normal duties. Repetitive laboratory tests using combinations of nine subjects and eleven different muffs have provided correction factors for the system relative to the ANSI S3.19-1974 psychoacoustic standard method for measurement of earmuff attenuation. The results of numerous field trials in terms of NRR, dBA reduction and octave-band attenuation values are presented for various muffs, occupations, and noise sources. These results are compared to those stated by the manufacturer.

4:11

II8. Exposure time versus noise level: A 3-, 5-, or 6-dB rule? David A. Bies and Colin H. Hansen (Department of Mechanical Engineering, University of Adelaide, G.P.O. Box 498, Adelaide, South Australia 5001, Australia)

The relationship between noise exposure and noise level represented in long-accepted published data [A. Glorig, in *Noise and Vibration Control*, edited by L. Beranek, Chap. 17, table 17.15; W. Burns and D. W. Robinson, *Hearing and Noise in Industry* (Her Majesty's Stationary Office, London, 1970), Appendices 10 and 11] is examined in a new way. It is shown that the data support the hypothesis that, in any situation involving a fixed time of exposure to a constant noise level, the same noise exposure (or hearing damage) is experienced if the noise level is increased by 6 dB and the exposure time is halved. This is contrary to regulations in Europe and the U.S.A. where the quantity 6 dB is replaced by 3 and 5 dB, respectively. An integral of pressure with time is consistent with the 6-dB observation, which also becomes the logical choice if the mechanism by which hearing damage accumulates is properly understood. A quantity identified as a hearing deterioration index (HDI) is introduced as a single scaling parameter for the data [D. A. Bies and C. H. Hansen, *Engineering Noise Control* (Unwin Hyman, London, 1988), Chap. 4].

WEDNESDAY AFTERNOON, 16 NOVEMBER 1988 — MOLOKAI ROOM, 2:00 TO 5:33 P.M.

Session JJ. Physical Acoustics IV: Role of Acoustics in High-Temperature Superconductors

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Chairman's Introduction—2:00

Invited Papers

2:05

JJ1. Anomalous ultrasound propagation in high-T_c superconductors. S. Bhattacharya (Corporate Research Science Laboratories, Exxon Research and Engineering Company, Annandale, NJ 08801)

Propagation of longitudinal and transverse ultrasound in high-T_c superconductors reveals unusual behavior at and below T_c. The superconducting order parameter is found to possess a large coupling to some volume-preserving shear distortions. Unlike in conventional superconductors, the behavior below T_c is inconsistent with a condensation of the carriers alone. The elastic modulus behavior below T_c is reminiscent of charge density wave/spin density wave transitions. The attenuation below T_c correlates with thermal transport and is related to either a strong phonon–phonon interaction or the Zener effect of intergranular heat transport in polycrystalline systems. Several anomalies in both the velocity and the damping are observed between T_c and room temperature, indicating the presence of other phase transitions of yet unknown origins. Magnetic field dependence of sound velocity shows fluctuation effects above T_c and a large stiffening below T_c probably related to an unusual flux-lattice dynamics. Implications of these results on the nature of superconductivity will be discussed.

2:25

JJ2. Flux lattices in the high-T_c superconductors. D. J. Bishop (AT&T Bell Laboratories, Murray Hill, NJ 07974)

Two types of experiments that provide information about the statics and dynamics of flux lattices in $YBa_2Cu_3O_7$ will be discussed. The first are magnetic decoration experiments in which the flux lines are decorated with magnetic particles. From these experiments, it has been learned that the flux lattice is triangular with the normal value for the flux quantum. In ultrasound and high-Q mechanical oscillator experiments flux lattice melting has been examined. These experiments show that the flux lattice melts well below Hc_2 into a vortex liquid state. These results suggest that the vortex solid and its melting is essentially two dimensional in character.

2:45

JJ3. Acoustic studies of single-crystal high-temperature superconductors. Brage Golding, W. H. Haemmerle, and L. F. Schneemeyer (AT&T Bell Laboratories, Murray Hill, NJ 07974)

The acoustic properties of single crystals of the high-temperature superconductor $YBa_2Cu_3O_7$ have been measured at temperatures between 0.1 and 300 K for frequencies near 10^3 and 10^9 Hz. In the GHz regime, longitudinal modes have been studied for propagation directions parallel and perpendicular to the *c* axis. At T_c, there is a discontinuity in the sound velocities and their temperature derivatives from which the anisotropic strain dependences of T_c are obtained. In the kHz regime, resonant excitation of flexural modes in thin reeds of $YBa_2Cu_3O_7$ crystals has permitted precise measurement of acoustic damping and dispersion. The temperature-dependent damping is characterized by at least five features associated with the relaxation of defects. At temperatures below 1 K, the velocity of sound is consistent with the presence of a broad "glasslike" distribution of tunneling modes.

3:05

JJ4. Optical phonons and electron–phonon interactions in high-T_c materials as probed by Raman scattering. M. V. Klein, S. L. Cooper, F. Slakey, B. G. Pazol, J. P. Rice, and D. M. Ginsberg (Department of Physics and Materials Research Laboratory, University of Illinois at Urbana–Champaign, 104 S. Goodwin Avenue, Urbana, IL 61801)

Raman scattering has played two roles in high T_c superconductivity studies: characterization and fundamental physical properties. After one has separated intrinsic spectra from those due to impurity phases, the physical properties of the intrinsic superconducting material may then be studied. In single crystals of $YBa_2Cu_3O_{7-\delta}$, the most interesting Raman-active mode is one at 330 cm^{-1} due to O atoms in the "planes." This mode has nearly pure B_{1g} tetragonal symmetry, even in the orthorhombic phase, and it shows strong softening below T_c. In detail, its Raman spectra reveal a Fano antiresonance with a continuum [Cooper *et al.*, Phys. Rev. B 37, 5920–5923 (1988)] of the same symmetry. An A_g mode at 118 cm^{-1} assigned primarily to Ba vibrations shows an even stronger antiresonance, but little softening. The continuum responsible for its antiresonance has A_g symmetry and a somewhat different shape from that of the B_{1g} continuum, but both Raman continua extend all the way to zero Raman frequency shift, implying linear low-frequency response. For $T \ll T_c$, the linear response is still present with five times reduced intensity, implying some form of gaplessness to the superconducting state. Efforts to rule out spurious surface effects will be described, and comparisons with results on materials such as $Bi_2Sr_2Ca_1Cu_2O_{8+\delta}$ will be made. [Work supported by NSF under DMR 8715103 and DMR 8612860.]

WEDNESDAY PM

3:25

JJ5. Ultrasonic studies of the roles of the electron–phonon interaction in high-T_c superconductors. Shoichi Mase and Yuuji Horie (Department of Physics, Kyushu University, Fukuoka, 812 Japan)

The role of the electron–phonon interaction in high-T_c superconductors $MBa_2Cu_3O_7$ (M = Y, Nd, and others), $(La_{1-x}Sr_x)_2CuO_4$, and $BaPb_{1-x}Bi_xO_3$ were studied by acoustic means. At scores of MHz, only $BaPb_{1-x}Bi_xO_3$ showed an appreciable electronic contribution to the sound attenuation, suggesting the largeness of the electron–phonon interaction. For higher T_c samples, no evidence of the electronic contribution was found, as can be predicted by the absence of the isotope effect in $YBa_2Cu_3O_7$. Structural instability was, however, investigated for some samples, and the energies of optical modes were estimated. Based on these experimental results and others, the phonon dispersion curves and the density of state curves $D(\epsilon)$ were calculated in agreement with inelastic neutron scattering data on $D(\epsilon)$. The calculated results show that the optical modes due to Cu–O and O–O bonds in $YBa_2Cu_3O_7$ provide large contributions to the electron–phonon interaction in view of the energy scale and symmetry properties of wavefunctions. There will be some discussion on what factors counterbalance these advantages. Using single crystals of $YBa_2Cu_3O_7$ and $NdBa_2Cu_3O_7$, measurements are being carried out to derive more accurate acoustic information.

3:45

JJ6. Resonant ultrasound measurements in single crystals of intrinsic and superconducting copper–oxygen plane compounds. Albert Migliori, Stuart E. Brown, Zachary Fisk, Eric T. Ahrens[a] (Los Alamos National Laboratory, Los Alamos, NM 87545), J. D. Maynard, and J. H. Mather[b] (Pennsylvania State University, University Park, PA 16802)

Resonant ultrasound measurements in very pure single crystals of La_2CuO_4 have been made. The material was prepared such that its resistivity at 300 K is extremely high (170 Ω cm), and untwinned crystals of about $0.1 \times 0.1 \times 0.1$ cm were used. Oxygen was then introduced, eventually leading to superconductivity. Here, data on the three longitudinal sound velocities and attenuations as a function of temperature for different oxygen dopings are presented and the effects of magnetic and superconducting interactions are discussed. All data were taken with a resonant ultrasound method developed by us in which the transducers are flexible and are less than 1% of the sample mass, minimizing transducer loading effects. [a] Work performed under the auspices of the U.S. Department of Energy. [b] This work was supported, in part, by NSF Grant DMR 8701-682 and the Office of Naval Research.

4:05

JJ7. Ultrasonic studies of sintered $YBa_2Cu_3O_{6.9}$ and La_2CuO_4 single crystals. Yuichi Okuda (Institute for Solid State Physics, University of Tokyo, Minato-ku, Tokyo, 106 Japan)

Single-phase sintered $YBa_2Cu_3O_{6.9}$ exhibits no anomaly in the attenuation around T_c (90 K). On the other hand, its sound velocity shows an anomalous increase just below T_c. This may come from lattice hardening due to the condensation of holes. To clarify the relation between this anomaly and the superconductivity, measurement under 8 Tesla was performed. Exactly the same anomaly was observed at 90 K and nothing happened around 72 K, which is the critical temperature under 8 Tesla. The hardening may not be directly related to the superconductivity of this material. An ultrasonic study of a single crystal of La_2CuO_4 is being made to investigate its antiferromagnetic property, which may be a key in understanding the new superconductivity. A small peak in the attenuation at T_N and a very small anomaly in the sound velocity were observed at 43 MHz. The result of a study on a single crystal of $YBa_2Cu_3O_{6.9}$ will also be presented at the conference.

4:25

JJ8. Anomalies in the elastic properties of the high-temperature superconductors. D. P. Almond, G. A. Saunders, E. F. Lambson, A. Al-Kheffaji, and M. Cankartaran (Schools of Materials Science and Physics, University of Bath, Bath BA2 7AY, United Kingdom)

Measurements will be presented of the ultrasonic wave velocity and attenuation in high-density polycrystalline samples of $YBa_2Cu_3O_7$ at temperatures between 4.2 and 300 K. Both longitudinal and shear wave data were found to be characterized by thermal hysteresis and run-to-run variations. At temperatures below T_c, the hysteresis disappeared, the elastic constants increased, and the attenuation fell. The higher temperature measurements were affected by sample annealing and thermal history and exhibited recovery effects. The pressure dependence of the elastic constants was found to be very large and also to exhibit hysteresis effects. These

phenomena are consistent with the suggestion that these ceramics are pseudoplastic solids in which the plasticity is attributed to the mobility of twin boundaries. Recent measurements of Bi–Ca–Sr–Cu–O superconductors will also be presented.

4:45–4:57

Break

Contributed Papers

4:57

JJ9. Ultrasonic studies of high-T_c superconductors. Yuuji Horie, Yuichiro Terashi, Hiroshi Fukuda, Masanori Hidaka, Takeshi Fukami, and Shoichi Mase (Department of Physics, Kyushu University, Fukuoka, 812 Japan)

High-T_c oxide superconductors [$MBa_2Cu_3O_7$ (M = Y and lanthanoids), $(La_{1-x}Sr_x)_2CuO_4$, $BaPb_{1-x}Bi_xO_3$, $x<0.35$] and their related insulating oxides [$MBa_2Cu_3O_6$, $BaPb_{1-x}Bi_xO_3$, $x>0.35$] were investigated by means of ultrasonic measurements. By using sound waves of $f_0 = 10$ MHz, we observed several peaks in the curves of the attenuation coefficient $\alpha(T)$ versus the temperature T for the superconducting samples. These peaks can be attributed to sound energy dissipation accompanied by excitations in optical phonons. However, there was no obvious anomaly near T_c except for $BaPb_{0.8}Bi_{0.2}O_3$ with $f_0 \sim 30$ MHz. The electronic contribution to $\alpha(T)$ was estimated to be very small in the Pippard mechanism of attenuation because of low electrical conductivity and a low f_0. In order to increase f_0, measurements with single-crystal samples having smaller resistivity are in progress. On the other hand, in the insulating samples, there are several peaks in the $\alpha(T)$ versus the T curves that can be assigned to some structural modulation. This assignment is supported by these piezoelectric and capacitance measurements. On the basis of these experimental results, the phonon contributions to high-T_c superconductivity are discussed.

5:09

JJ10. Ultrasonic attenuation and velocity investigation of sinter-forged superconducting $YBa_2Cu_3O_7$. M. Levy, B. K. Sarma, M.-F. Xu (Physics Department, University of Wisconsin—Milwaukee, Milwaukee, WI 53201), S. Adenwalla, Z. Zhao, Q. Robinson, and J. B. Ketterson (Physics and Astronomy, and MRC, Northwestern University, Evanston, IL 60208)

Sinter-forged $YBa_2Cu_3O_7$ exhibits rotational symmetry about the forging axis with the c axis of 80% of the crystallites aligned within 20° of the forging axis. Attenuation measurements as a function of temperature of longitudinal waves propagating perpendicular to the forging axis exhibit three broad relative maxima at 70, 180, and 250 K. Attenuation measurements with either longitudinal or transverse waves propagating along the forging axis exhibit only one maximum at about 180 K. The absence of the other two maxima along this orientation indicates that they could be produced by interaction mechanisms that may be associated with the Cu–O planes, while the 180-K maximum may be produced by an isotropic interaction mechanism. Sound velocity measurements show the sinter-forged material to be elastically anisotropic. The temperature dependence of the sound velocity shows distinct hysteresis curves that flatten out if the sample is not warmed above 250 K. Upon cooling, lattice softening is observed to start at temperatures much higher than T_c, and to stop below T_c. [Work supported by ONR at the University of Wisconsin—Milwaukee and by NSF at Northwestern University.]

5:21

JJ11. Can you hear the elastic tensor? William M. Visscher (Theoretical Division, MS B262, Los Alamos National Laboratory, Los Alamos, NM 89745)

The measured response of an object of known size and shape composed of a homogeneous but anisotropic elastic material as a function of frequency of excitation is, in principle, enough to determine the linear elastic constants, 21 of them in the most general case. A method for computing the resonant frequencies is explained and illustrated for an object of simple geometry ($a \times b \times c$ rectangular parallelepiped) with free or clamped faces of a material whose elastic modulus tensor reflects the symmetry of an orthorhombic lattice (nine independent constants). The method is a variant of MOOT [method of optimal truncation, see J. L. Opsal and W. M. Visscher, J. Appl. Phys. **58**, 1102 (1985)], which is a boundary residual method developed to compute elastic wave scattering. The calculation was undertaken as a part of the analysis of measurements of ultrasonic response of single-crystal, high-temperature superconductor materials (reported elsewhere at this meeting). Results will be presented, and the practicality of this approach to determine unknown elastic constants will be discussed. [Work supported by USDOE.]

Session KK. Speech Communication VII: Production, Part C (Poster Session)

Catherine T. Best, Cochairman
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Contributed Papers

All posters will be displayed from 2:00 to 4:00 p.m. To allow contributors the opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 2:00 to 3:00 p.m. and contributors of even-numbered papers will be at their posters from 3:00 to 4:00 p.m.

KK1. A study on the formant analysis of Korean monophthongs and their resonance effects in vocal tract. Hyun Jae Shin and Suk Wang Yoon (Acoustics Research Laboratory, Department of Physics, Sung Kyun Kwan University, Suwon, Republic of Korea 440-746)

Twelve Korean monophthongs pronounced by five male vocal musicians with five fundamental frequencies were studied by formant analysis. Fundamental frequencies and their harmonics were considered as the parameters of analysis. This study shows that the first and the second formants are characterized by the resonance of the cavities of pharynx and mouth, respectively. The lip-rounding effect decreases the second formant frequency. The phonemes of /a, ɑ/, /e, ɛ/, and /ə, ʌ/ were not distinguished well in this formant analysis for Korean.

KK2. Acoustic correlates of apical and laminal articulations. Sarah N. Dart (Phonetics Laboratory, Department of Linguistics, UCLA, Los Angeles, CA 90024)

Several indigenous languages of the western United States have a phonemic contrast between segments articulated with the apex of the tongue up (apical) and those articulated with the apex down behind the lower teeth and the contact on the palate made instead with the blade of the tongue (laminal). This difference in apical and laminal articulation is also seen in individual variation in languages without such a phonemic distinction. In the present study, palatograms and linguagrams with synchronous audio recordings were taken of a number of speakers of French, American English, and several different Native American languages, illustrating apical and laminal articulations in languages both with and without the pertinent phonemic distinction. The segments studied include /t,d,n,l,s,z/. This paper reports on the acoustic correlates of the articulatory differences found within each language as well as some cross-language comparisons. [Work supported by NSF Grant BNS 8704361.]

KK3. Coarticulation effects in Japanese velar stop consonants: Observations with dynamic velography. Noriko Suzuki (Department of Oral and Maxillofacial Surgery, Showa University, Tokyo, Japan and Massachusetts Institute of Technology, Room 36-513, Cambridge, MA 02139) and Ken-ichi Michi (Department of Oral and Maxillofacial Surgery, Showa University, Tokyo, Japan)

The positioning of the tongue body is strongly influenced by coarticulation effects. The tongue body contact position for /k/ and /g/ varies especially widely, moving in the direction demanded by the surrounding phonemes. Coarticulation effects of post-consonantal vowels on the place of tongue body contact for Japanese velar stop consonants were observed by dynamic velography. The dynamic velography system, developed by the authors is a 36-electrode electropalatographic technique for observing lingual contact with the soft palate. The resulting velograms can be compared with spectrograms made from simultaneous recordings of speech. Three Japanese male subjects produced VCV utterances with stress on the second syllable. The velogram patterns for [aka], [ako], [aga], [ago] were similar, with contact located at the posterior part of the soft palate. Contact for [ake], [aki], [age], [agi] was located more anteriorly. Thus, as suggested by previous findings, the tongue contact position for Japanese /k/ and /g/ is strongly influenced by the vowels following it.

KK4. Effects of variation in speaking rate, loudness, and vocalic context on linguapalatal contact patterns in Hindi sibilants. R. Prakash Dixit (Department of Speech: Communication Disorders, Louisiana State University, Baton Rouge, LA 70803) and James E. Flege (Department of Biocommunication, University of Alabama at Birmingham, Birmingham, AL 35294)

A 96-channel electropalatograph was used to monitor normal, fast, and loud production of sibilants in the nonsense words /bisíb/, /basáb/, /busúb/ and /biš́ib/, /bašáb/, /bušúb/ spoken in a carrier phrase by a native speaker of Hindi. Results showed the following. (1) Groove width (GW, in number of uncontacted sensors $\times 2$ mm) was considerably narrower for /s/ (4.64 mm) than /š/ (9.26 mm). (2) GW for /s/ decreased from normal to fast to loud speech; just the opposite was true for /š/. (3) For both /s/ and /š/, GW was narrower in the context of /a/ than /i/ or /u/; in the latter two, it was narrower in the /u/ context. (4) Anterior–posterior location (APL) of the groove (in number of contacted rows) occurred generally at the second and third rows for /s/ and the fourth and fifth rows for /š/. Consequently, the center of the groove for /s/ was 4 mm anterior to that for /š/ (i.e., the difference in the number of rows $\times 2$ mm). (5) APL of the groove for /s/ was more anterior in loud than normal or fast speech; in the latter two, it was more anterior in normal speech. (6) Groove APL for /s/ was more anterior in the context of /a/ than /i/ or /u/; in the latter two, it was more anterior in the context of /i/. (7) For /š/, groove APL was more posterior in fast than normal or loud speech; in the latter two, it was more posterior in loud speech. (8) Groove APL for /š/ shifted more posteriorly from /i/ to /a/ to /u/ context. [Work supported by NIH Grant NS20572.]

KK5. Acoustic differences correlated with derivational history. M. Peet (The MITRE Corporation, 7525 Colshire Drive, McLean, VA 22102-3481) and M. Withgott (Xerox Palo Alto Research Center, 3333 Coyote Hill Road, Palo Alto, CA 94304)

In a study of American English palatals, acoustic characteristics of allophones with different derivational histories were compared. In examples such as "toss sheepskins" versus "josh Sheila," palatalization was found to be complete, as observed in a spectral analysis, and no rearticulation was found. However, the š derived from a sequence of two š's was observed to exhibit longer duration values than the s from an sš sequence, in keeping with the underlying, intrinsic segment duration values (cf. also, Zue and Shattuck-Hufnagel, 1980). The results suggest that a production/perception theory should model the underlying segment along with the assimilation process. The data were collected from a corpus of 60 sentences each read by eight adult male speakers. [Work supported, in part, by DARPA-ISTO.]

KK6. Aerodynamic constraints on language history: The case of bilabial trills. Ian Maddieson (Phonetics Laboratory, Department of Linguistics, UCLA, Los Angeles CA 90024-1543)

Bilabial trills, transcribed [B], are markedly rare as speech sounds in the world's languages, even though Catford (1977) claims that they are the easiest type of trill to produce. With one exception (Liangshang Yi), all languages known to have bilabial trills developed them in a highly restricted environment consisting (historically) of a sequence of a voiced bilabial nasal, a voiced bilabial stop, and a high rounded vowel, e.g., [mbu] → [mBu]. In many cases, the trill remains allophonic (e.g., Na?ahai, Windua), but some languages have undergone a restructuring resulting in phonemic bilabial trills (e.g., Kurti, Atchin). The limitation to this environment suggests that aerodynamic conditions normally only satisfied in speech by this sequence are required for labial trilling to develop. The essentials would seem to be: (1) transglottal airflow without intraoral pressure buildup, permitted by the nasal escape in [m] and not significantly affected by the very brief closure typically found for a homorganic post-nasal stop; (2) subsequent maintenance of bilabial closure without nasal flow, providing for an oral release; and (3) a following vowel with a target position for the lips of narrow aperture. Since this target lip position requires only a small movement from closure, articulator movement is slow [Kent and Moll (1972)]; hence, there is a period of time during which the lips remain close enough together for Bernouilli forces, in the absence of increased intraoral pressure, to reclose them and initiate trilling. A simplified quantitative model of this process will be presented, together with speculations on why bilabial trills do not otherwise occur linguistically. [Research supported by NSF Grant BNS 87-20098.]

KK7. An acoustic analysis of English liquids uttered by the Japanese and native speakers of English. Hirotake Nakashima, Yukihiro Nakayama, and Charles McHugh (Faculty of International Language and Culture, Setsunan University, 17-8 Ikedanakamachi, Neyagawa, 572 Japan)

This research deals with the production of English liquids through an acoustic analysis of speech samples uttered by Japanese college students, advanced Japanese learners of English (college teachers of English), and native speakers of English 13–18 years of age who live in Japan. The speech samples are 64 English words with liquids in initial position. These speech samples were analyzed by the autocorrelation method of linear prediction to estimate the formant frequencies every 10 ms using an analysis window of 20-ms length. The second and third formant frequencies were extracted for each of the liquids. The results showed that: (1) There is a significant difference in the formant values of each liquid between males and females; (2) the formant values of each liquid for advanced Japanese learners of English coincide with those for native speakers of English; and (3) although the values of $F2$ and $F3$ of /r/ and /l/ uttered by Japanese students overlapped with each other on the $F2$–$F3$ plane, these separated clearly after pronunciation training.

KK8. A comparison of evaluations by American and Japanese listeners of English spoken by Japanese speakers. Hiroshi Suzuki (English Department, College of Arts and Sciences, University of Tokyo, 3-8-1 Komaba, Meguro-ku, Tokyo, 153 Japan) and Ghen Ohyama (Hearing and Speech Perception Department, ATR Auditory and Visual Perception Research Laboratories, Twin 21 MID Tower, 2-1-16 Shiromi, Higashi-ku, Osaka, 540 Japan)

By means of the PARCOR analysis and synthesis technique, various combinations of the three prosodic features, i.e., the duration of each sound, pitch change, and the intensity change of an English sentence uttered by Japanese speakers in typically Japanese fashion were replaced with the same combinations of the prosodic features of the same English sentence read by an American. A group of Americans and a group of Japanese listened to the recording of the modified utterances and judged their English acceptability level, or "Englishness." For the Americans, pitch seems to play a more important role in such judgments than duration and intensity, while, for the Japanese, pitch is far more important than duration, and intensity seems least important.

KK9. Evaluation of English pronunciation based on the static and dynamic spectral characteristics of words spoken by Japanese. Hiroshi Hamada and Ryohei Nakatsu (NTT Human Interface Laboratories, 1-2356 Take, Yokosuka, 238-03 Japan)

To develop an English pronunciation training system, a method is proposed for evaluating the pronunciation ability of Japanese speakers. The quality of English pronunciation is assumed to be determined by the static characteristics of phonetic spectra, the dynamic structure of spectrum sequences, and prosodic characteristics of utterances. Since it is difficult to evaluate these factors directly, evaluation is achieved by comparing English words uttered by a Japanese speaker with those uttered by a native speaker using speech recognition techniques. The static characteristics of phonemes are evaluated by measuring the stability of mapping functions that adapt phonetic spectra of Japanese speakers to those of native speakers. The mapping functions are obtained by speaker adaptation through vector quantization. Evaluation values for the dynamic spectral structure are defined by the DTW matching distance between words spoken by Japanese and those spoken by native speakers. Although prosodic characteristics are not considered, preliminary experiments show that the evaluation results obtained by the proposed method have a good correspondence with human judgments of pronunciation quality.

WEDNESDAY PM

KK10. Acoustic measurements of induced slips of the tongue in children and adults. Bruce L. Smith (2299 Sheridan Road, Department of Communication Sciences and Disorders, Northwestern University, Evanston, IL 60208)

Although many studies of slips of the tongue have been conducted with adults, very few investigations have considered children's slips of the tongue. Furthermore, because spontaneous slips of the tongue typically are not captured with tape recordings, very few acoustic studies of slips of the tongue have been conducted with any subjects. The present study utilized an elicitation technique to induce slips of the tongue in a group of 5-year-old children and a group of adults. Subjects repeated short tongue-twister phrases (e.g., Swiss wristwatch shop), as well as control phrases that were easier to produce (e.g., Swiss chocolate store). The types of errors that subjects made (substitutions, dysfluencies, etc.) were computed, and acoustic measurements of certain segments were made. One observation made from the acoustic data was that segment durations produced by both the children and the adults were 30%–40% longer for the tongue twisters versus the control phrases, even when productions were correct in both conditions. Additional findings and their implications concerning speech production will be presented.

KK11. Infants' vocalizations in mother–infant interaction. Yoko Shimura (Department of Education, Saitama University, 255 Shimo-okubo, Urawa, 338 Japan), Satoshi Imaizumi (RILP, Faculty of Medicine, University of Tokyo, 7-3-1 Hongo, Bunkyo-ku, Tokyo, 113 Japan), Tamiko Ichijima (Sophia University, 7-1 Kioicho, Chiyoda-ku, Tokyo, 102 Japan), and Kozue Saito (Kokugakuin University, 4-10-28 Higashi, Shibuya-ku, Tokyo, 150 Japan)

The vocal behavior of young infants was investigated to clarify the process of speech acquisition. Acoustic and picture analyses were carried out for audio and video recordings of 26 young infants (aged 4 to 19 months) being addressed by their mothers, unfamiliar adults, or voices presented through a loudspeaker. The results were the following. (1) Mothers tended to use a wider range of F_0 than those observed in their speech toward adults. (2) The F_0 upper limits in the mothers' speech was closely related to the F_0 of their infants' voices. (3) Young infants produced various voice qualities, which could be characterized by the richness of their subharmonics, the richness of noise, or a sudden change in F_0. (4) Not only the qualities of the infants' voices, but also their facial expressions and looking behavior seemed to change according to how and by whom they were being addressed. [Work supported by Toyota Foundation.]

KK12. Phonologically motivated substitutions in a 20–22 month old's imitations of intervocalic alveolar stops. Catherine T. Best and Deborah Wilkenfeld (Haskins Laboratories, 270 Crown Street, New Haven, CT 06510)

Systematic deviations between early word productions and their adult targets reflect not only children's articulatory limitations, but also their phonological development. However, research in child phonology has overlooked an important source of information by excluding from analysis any direct imitations of adult words. Consistent with observations about syntactic and cognitive development, spoken imitations should be systematically modified by the child's phonological system. Imitation studies should thus reveal more about the sophistication of the early phonological system by minimizing demands on memory and lexical access, by allowing presentation of unfamiliar or nonsense words, and by permitting comparison of the child's productions with the actual phonetic and acoustic properties of the adult targets. The present research found remarkable phonological sophistication in a toddler's production of intervocalic stops, based on phonetic and acoustic analyses of her imitations of a set of phonetically reduced adult targets at 20–22 months of age. The adult targets were disyllabic words and nonwords containing /d/ or /t/ preceding ⟨-er⟩, ⟨-le⟩, or ⟨-en⟩, in which the stops were realized as the restricted phonetic variants [ɾ], [dn], or [ʔ], (e.g., respectively, ⟨wider⟩ or ⟨whiter⟩; ⟨widen⟩; ⟨whiten⟩). Although alveolar stops are not normally produced in this phonetic environment, the child consistently substituted them for the more restricted variants found in the adult targets. The child's phonology also distinguished /t/ and /d/. The findings indicate a rule like that in the adult grammar which determines how these phonemes will be realized in highly specified phonetic contexts and, consequently, how phonetically disparate forms are related in an abstract category. The child's failure to exactly imitate the target utterances and the systematicity of her deviations further argue that the targets were, in fact, filtered through the child's phonology before she produced them. [Work supported by NIH.]

KK13. Two-stage adult acquisition of intonational contours. Eric Keller (Department of Linguistics, Université du Québec à Montréal and Centre de recherche, CHCN, 4565 Queen Mary, Montreal, Quebec H3W 1W5, Canada)

The precise process of the acquisition of new speech motor patterns in adults has not been examined extensively. In particular, it is unclear how and in which order control over an unfamiliar motor pattern is acquired. The acquisition by two French-language speakers of unfamiliar (Chinese-like) intonational contours overlaid on the second syllable of /aCa/ nonsense stimuli was studied. Stimuli were recorded from a native speaker of Beijing Chinese. Thirteen hundred $F0$ measurements of responses obtained in nine training sessions, spread over 3 weeks, were evaluated for three types of learning: (1) accuracy (coefficients of variation over multiple attempts), (2) proximity to target frequency (Hz), and (3) contour fidelity (similarity between target and response contours, Hz). Results showed two acquisitional phases: A rapid, major improvement on measures (1) and (2), evident over the rest of the training period. There was a significant positive linear correlation between measures (1) and (2). These data support notions of both rapid learning and slow fixation and are thus in support of the concept of stored speech motor control patterns. This contradicts current motor theories that advocate new trajectory calculation for every movement, even in the case of well-established motor patterns. [Work supported by NSERC, Canada.]

KK14. What do mimics do when they imitate a voice? George Papcun (Computing Division and CNLS, Los Alamos National Laboratory, Los Alamos, NM 87545)

Imitations by both professional and amateur mimics were studied to determine what similarities are achieved between the imitated voice and the imitation thereof. A wide variety of characteristics was approximated, including the following: mean $F0$, $F1$, and $F2$; frequency contours of $F0$, $F1$, and $F2$; degree of nasalization (but not frequency of nasal formants); speech rate and dynamics, including timing, attack, and release characteristics. Contours of $F0$, $F1$, and $F2$ were often matched accurately, even when their absolute frequencies differed considerably from those of the original. Specific images of words and phrases were used, as well as general phonetic characteristics. Imitators tended to concentrate on imitating unusual characteristics of a voice, rather than attempting to imitate all characteristics equally. This observation may be formalized as a model according to which the importance of a parameter is nonlinearly related to the extent to which it diverges from its mean population value. Professional mimics exaggerated the distinctive characteristics of voices and thus may be considered caricaturists rather than mimics, *per se*.

KK15. Relations between formant slopes and speech intelligibility in neurologically impaired talkers. R. D. Kent, G. Weismer, J. F. Kent, R. E. Martin, and J. C. Rosenbek (Department of Communicative Disorders, University of Wisconsin–Madison, 1975 Willow Drive, Madison, WI 53706)

The relation between $F1$ and $F2$ slopes measured from monosyllabic words and speech intelligibility scores for these words was studied for 25 men with amyotrophic lateral sclerosis. The formant slopes were estimated from digitized formant tracks for each test word. The intelligibility scores were computed as the percentage of words correctly identified for each talker by a listener panel of ten young women. The intelligibility scores for the group of 25 talkers ranged from 41%–99%. The correlation coefficient computed between the $F2$ slopes and the intelligibility scores was 0.76. The $F1$ slope was not as predictive of intelligibility, perhaps because the dysarthric talkers compensated for impaired tongue control with large jaw movements (and, hence, large $F1$ shifts). It is concluded that $F2$ slope, a dynamic measure of acoustic structure, bears a moderately high correlation with intelligibility in these subjects. Examples of formant patterns are shown for different degrees of speech impairment. [Work supported, in part, by NINCDS.]

KK16. Articulatory dynamics of deaf speakers during plosive production: Aerodynamic and kinematic evidence. James J. Mahshie and Pradeep Yadev (Gallaudet University, Department of Audiology and Speech, Speech Production Laboratory, Washington, DC 20002)

The production of speech requires precise movement and timing of the articulators to accurately valve the egressive airstream. While evident that deaf speakers often experience difficulty controlling such valves for speech production, the nature of such valving difficulties remains elusive. Previous aerodynamic studies have been primarily descriptive, characterizing steady-state aspects of speech production. The present study described and compared dynamic aspects of deaf and normal-hearing speakers' speech production using aerodynamic and kinematic measures. Five profoundly deaf and two normal-hearing young adults produced multiple tokens of voiced and voiceless plosive segments in varied vowel contexts, while simultaneous measures were obtained of oral and nasal airflow, oral pressure, and electroglottograph and acoustic signals. Conductance of oral and nasal vocal tract constrictions were used to describe and compare the magnitude and relative timing of articulatory gestures for deaf and hearing subjects. Computer simulation of the supraglottal air pressure and airflow waveforms was used to further investigate how these waveform variations may be related to articulatory maneuvers. Implications for speech training will be discussed. [Research supported by the Whitaker Foundation.]

KK17. Quantitative evaluation of hypernasality in cleft palate patients. Ryuta Kataoka, Koji Takahashi, Yukari Yamashita, Satoko Imai, Ken-ichi Michi (Showa University, 2-1-1 Kitasenzoku, Ohta-ku, Tokyo, 145 Japan), Kaoru Okabe, Hareo Hamada, and Tanetoshi Miura (Tokyo Denki University, 2-2 Kanda-Nishiki-cho, Chiyoda-ku, Tokyo, 101 Japan)

To quantitatively evaluate hypernasality in cleft palate patients, the Japanese vowel /i/ pronounced by six cleft palate patients and four normal children (controls) of similar ages was analyzed acoustically by cepstrum analysis. Spectrum envelopes obtained by the cepstrum method were evaluated every 1/3 octave to obtain the mean level in each band. Ten listeners evaluated a speech sample from each subject for degree of nasality on an equal interval scale ranging from 0 (no nasality) to 4 (strongest nasality). Two factors were obtained from the factor analysis of the judged scores. The first factor, which accounted for 77% of the total variance, was the consensus perception of nasality. The second factor, which accounted for 9%, was the difference among the individual listeners. The levels in two 1/3 octave bands were highly correlated with the first factor. The central frequencies of these two bands were 1 and 5 kHz.

KK18. Acoustic–phonetic analysis of normal, loud, and Lombard speech in simulated cockpit conditions. Bill J. Stanton (Department of Electrical Engineering, U. S. Air Force Academy, Colorado Springs, CO 80840-5851), George D. Allen (Department of Audiology and Speech Sciences, Purdue University, West Lafayette, IN 47907), and Leah H. Jamieson (School of Electrical Engineering, Purdue University, West Lafayette, IN 47907)

It has long been recognized that raising one's voice causes perceptible changes in the speech signal beyond mere increase in overall intensity. This paper reports the results of an extensive analysis of speech produced in a simulated fighter cockpit environment, with helmet and oxygen mask in place. Eight talkers each produced 56 utterances under three conditions: (1) normal, (2) loud (nominally 10 dB above normal), and (3) Lombard (evoked by 90 dB of pink noise played through a headset). A total of 17 671 phonemes were hand marked for analysis using 18 acoustic features (ten frequency bands, spectral COG, low- and high-frequency spectral tilt, $F0$, $F1$–3, and duration). For most speakers, both loud and Lombard conditions showed the following shifts in comparison with the normal condition: (1) For vowels and sonorant consonants, lower (0–500 Hz) and higher (5–8 kHz) frequency bands lost energy relative to the mid (1–4 kHz) frequencies; (2) also for vowels and sonorants, $F0$, $F1$, and spectral COG all rose; and (3) for fricatives, affricates, and voiceless stops, lower (0–3 kHz) frequencies lost energy relative to higher (4–8 kHz) frequencies. There was much variation in these effects among talkers. [Work supported by Air Force Institute of Technology.]

KK19. Fricative consonants: Comparisons between human and mechanical-model production. Christine H. Shadle (Department of Electronics and Computer Science, University of Southampton, Southampton SO9 5NH, England)

In an earlier paper [C. Shadle, J. Acoust. Soc. Am. Suppl. 1 **82**, S15 (1987)], the derivation of source parameters for fricative consonants from mechanical models of the vocal tract was discussed. Source location and source spectrum at a range of airflows were measured for the fricatives /ʃ, ç, x/, and the separation of these fricatives into two distinct acoustic types was proposed. This paper presents further analysis of these data. A collapsing of the source spectra into a single curve for each fricative, parametrized by flow rate, is proposed. These collapsed source characteristics were used in a frequency-domain model of the vocal tract [P. Badin and G. Fant, STL-QPSR **2-3**, 53–108 (1984)] to predict the far-field sound generated. These predicted spectra were compared to the sound produced by the mechanical models and by Fant's subject on which the models were based. Some discrepancies between experiment and theory are apparent; these are due to the lack of higher modes in the computer model, some slight anatomical inaccuracies in the mechanical model in the region of the constriction, and probably also to spatial distribution of the source for /ç,x/. Aside from these discrepancies, the comparison demonstrates the validity and usefulness of the mechanical-model data.

Session LL. Structural Acoustics and Vibration IV: Flow-Induced Sound and Vibration

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Chairman's Introduction—2:00

Invited Papers

2:05

LL1. The basics of flow noise. Eugen J. Skudrzyk (Applied Research Laboratory, The Pennsylvania State University and the Physics Department, State College, PA 16804)

The flow noise nearfield that is recorded by a flush-mounted hydrophone is generated by a "Bernoulli" effect. Because the pressure within the eddies is less than that in the fluid outside, fluid is injected into the eddies and thrown against the wall. The kinetic pressure is then recorded as nearfield noise. The nearfield sensitivity of a hydrophone seems to depend greatly on its shape and size. But a closer investigation shows that its sensitivity is inversely proportional to its extension in the direction of the flow and is approximately independent of its width. At high frequencies, the radiation field dominates over the nearfield. The radiation field is described by the Lighthill equation. Its left-hand side is identical with the wave equation for an inviscid fluid; on the right, turbulence-generating effects are accounted for by a known "forcing function." For a point force, the solution is an image source supplemented by a thin layer of evanescent shear waves along the boundary. A similar solution is obtained for multipoles and for turbulence. There is no layer of sound-radiating dipoles nor a resonance board effect, as is proved by performing the computation in terms of the Fourier amplitudes.

2:25

LL2. Generation of discrete frequency noise due to periodic vortex shedding from a streamlined body. Tohru Fukano (Department of Mechanical Engineering, Kyushu University, 6-10-1 Hakozaki, Higashi-ku, Fukuoka, 812 Japan)

The mechanism of discrete frequency noise generated from a flat plate with a sharp trailing edge immersed in a uniform oncoming flow was investigated experimentally. The results show that discrete frequency noise is caused by the periodic shedding of Karman vortices, and the necessary conditions of its generation are summarized as follows: (1) There must be a dead flow region with a definite area of a sufficiently large scale. (2) The shear layers starting from the two separation points near the trailing edge of the body must be very strong on either or both sides of the plate. A flow model was introduced to estimate the sound-pressure level theoretically and was verified experimentally as useful. The characteristics of Karman vortex shedding from a rotating blade were also examined.

2:45

LL3. Control of Karman-vortex sound by two crossing cylinders. Yoshiyuki Maruta (The 3rd Laboratory, EBARA Research Company, Ltd., 4-2-1, Honfujisawa, Fujisawa, 251 Japan) and Shoji Suzuki (Faculty of Engineering, Hosei University, 3-7-2 Kajino-cho, Koganei, 184 Japan)

The pure tone generated from a cylinder's wake with a Karman vortex in the airflow is the basic phenomenon of aerodynamic sound and is also the sound that is often generated by flows of wind around constructions and the inner flow of turbomachinery. By putting another crossing cylinder into contact with the first, the Karman-vortex sound becomes quieter than for a single cylinder [Maruta *et al.*, Proc. Internoise **87**, 481–484

(1987)]. This phenomenon was investigated experimentally by changing the diameters, space, and crossing angle between the two cylinders. For cylinders with the same diameter, normal crossing was more effective for this sound reduction. For the ones with different diameters, inclined crossing was more effective with space less than 1.5 times the upstream cylinder's diameter. The sound reduction results from the fact that the spanwise coherence of the Karman vortex from the upstream cylinder is prevented by crossing with other cylinders. The Karman-vortex sound can be controlled by the optimum condition of two crossing cylinders.

3:05

LL4. The response of plates to TBL excitation: High versus low wavenumber effects. Nathan C. Martin (BBN Systems and Technology Corporation, 10 Moulton Street, Cambridge, MA 02139)

Structural response to turbulent boundary layer (TBL) wall pressure fluctuations has been recognized as an important consideration in the design of aircraft and ships for many years. Unlike most sources of structural response, the TBL generates fluctuating pressure components over a wide range of spatial (i.e., wavenumber) as well as temporal (i.e., frequency) scales. In the frequency range of interest for many applications, the dominant spatial components of TBL excitation lie at relatively high wavenumbers compared to the major acceptance regions of resonant structural modes. Chandiramani [J. Acoust. Soc. Am. **61**, 1460 (1977)] has shown that the low wavenumber components of TBL excitation dominate the response of plates with simply supported boundary conditions when certain assumptions are made about the relative levels of TBL high and low wavenumber content. The purpose of this paper is to re-examine the subject of the relative importance of high and low wavenumber contributions to TBL-excited plate response as influenced by potential variables such as the various models of TBL wavenumber content and the nature of the plate boundary conditions.

3:25

LL5. Design considerations for reducing propeller cavitation noise. Tetsuji Hoshino, Takao Sasajima, and Akira Oshima (Ship Hydrodynamics Laboratory, Nagasaki Research & Development Center, Mitsubishi Heavy Industries, Ltd., 3-48 Bunkyo-machi, Nagasaki, 852 Japan)

In designing naval or oceanographic ships, the reduction of underwater noise radiated from the ship is of primary importance to secure the reliable operation of acoustic instruments. Among various hydrodynamic and mechanical noise sources, propeller cavitation is considered to be most harmful for acoustic survey operation. A design for low-noise propellers and design examples for oceanographic research ships are presented. The cavitation patterns and radiated noise characteristics of the propellers thus designed were investigated by model tests in a cavitation tunnel. It was shown that the radiated noise level from the propeller was sufficiently low in the model scale. This was further confirmed by radiated noise measurements on the propeller at full scale. The comparison of the full-scale noise data with those estimated from the model scale noise data showed that the noise measurements in a cavitation tunnel are useful in evaluating the design of a low-noise propeller.

Contributed Papers

3:45

LL6. Laser Doppler vibrometer technique for measuring surface waves produced on submerged elastomeric layers. Timothy E. McDevitt and Alan D. Stuart (Applied Research Laboratory, The Pennsylvania State University, P. O. Box 30, State College, PA 16804)

A dual-channel, combination vibrometer/velocimeter laser Doppler system has been developed to measure the surface waves produced on a submerged elastomeric layer excited mechanically or by a fluid flow. This nonintrusive laser system permits velocity measurements to be made both normal and in the plane of the layer's surface without distorting its vibratory field. Since two laser systems are employed, measurements can be made simultaneously in two locations: permitting the spatial coherence of the surface disturbance to be assessed and transfer functions between source and response to be determined. This paper will present the results obtained for a long sample of elastomeric material submerged in water, and driven both longitudinally and in flexure to simulate typical in-plane and transverse motion of its surface. [Work supported by ONR.]

3:57

LL7. Acoustic concepts in nonuniform, steady potential flows. L. M. B. C. Campos (Instituto Superior Técnico, 1096 Lisbon, Portugal)

The concepts of Doppler factor, local Doppler-shifted frequency, Blikhintsev wave invariant, and group velocity, which are well known for sound in a uniformly moving medium, and in the ray approximation to the acoustics of nonuniform flows, are generalized to waves of arbitrary frequency in a steady, nonuniform potential flow of arbitrary Mach number. The generalized concepts coincide with the usual Doppler factor, local frequency, wave invariant, and group velocity, in the ray approximation, and their definition is made unique by the requirement that, outside the ray approximation, the following relations remain valid: (i) The generalized wave invariant remains an adiabatic invariant, in the sense that it equals the total (kinetic plus compression) energy divided by local frequency; (ii) the latter is related to the wave frequency through the generalized Doppler factor; (iii) the energy velocity is the ratio of energy flux to energy density. All these concepts depend on convection and energy partition factors, which reduce to unity in the ray limit. These concepts are introduced on the basis of the acoustic energy equation, which can be derived from a variational principle, which also yields wave equations.

4:09

LL8. Tip vortex sound. Jonathan L. Gershfeld and Edward J. Skiko (David Taylor Research Center, Code 1944, Bethesda, MD 20084-5000)

The aerodynamic sound generated by the convection of a tip vortex past a trailing edge is examined. For angles of attack greater than ± 5 deg, a vortex is generated that produces a high-frequency fluctuating pressure field several hundred hertz in bandwidth, and as much as 5 dB above the boundary layer wall pressure field, centered at roughly twice the frequency of vortex shedding due to Helmholtz wake instabilities. Sound is generated as the vortex-induced pressure field is scattered by the trailing edge. The increase in the acoustic source level, quantified by fluctuating boundary layer surface pressure statistics both near the trailing edge of the foil and under the tip vortex, corresponds in level and frequency to the increase in measured radiated noise. The wall pressure field due to tip vortex flow is superimposed on the boundary layer pressure field and apparently does not alter the wall pressure field at other frequencies. The directivity of the radiated tip vortex sound shows a reduction in the sound in the plane of the foil around the tip. This suggests that a quarter plane Green's function is required to describe the radiated noise. [Work supported by O.N.R.]

4:21

LL9. Wall pressure fluctuations in the transition region. Thomas A. Galib (Naval Underwater Systems Center, Code 8122, Newport, RI 02841-5047)

Wall pressure fluctuations on an axisymmetric body of revolution were measured with piezoelectric pressure transducers. Spectral analysis of the data showed discrete frequency bands of the pressure fluctuations corresponding to predicted Tollmien–Schlichting disturbance frequencies in the transitional boundary layer. Nondimensional power spectra showed that the peaks of the disturbance frequencies (for three free-stream velocities) collapsed to a single Strouhal number. Further manipulation of this result yielded an expression for the Tollmien–Schlichting frequencies in terms of a constant and the displacement thickness Reynolds number. These results were for a near-zero pressure gradient. They were later reproduced in an adverse pressure gradient for a variety of Reynold's numbers.

4:33

LL10. Vibroacoustic analysis of a space shuttle payload. Y. Albert Lee (D62-18, B104, Lockheed Missiles and Space Co., Sunnyvale, CA 94086)

A Vibroacoustic payload environment prediction system (VAPEPS) has been developed to perform statistical energy analysis of spacecraft under acoustic excitation. VAPEPS was employed to analyze the random vibration response of the NASA Office of Space Science-1 (OSS-1) Payload in the space shuttle cargo bay acoustic environment. The OSS-1 payload consists of various experimental hardwares mounted on a pallet. The pallet is constructed of a framework covered with face panels that is responsive to the acoustic pressure field. The entire pallet structure was modeled as an equivalent plate with the same dimension, stiffness, surface mass density, and longitudinal wavespeed. Detailed models were also constructed of the individual panels of honeycomb construction, which make up the overall face panels. The predicted acceleration power spectral density in third octave bands was compared with experimental data measured at system level acoustic test in a reverberant chamber. The agreement is good.

4:45

LL11. Noise radiation from a jet impinging on a flat plate. Jianping Shen and William C. Meecham (Department of Mechanical, Aerospace, and Nuclear Engineering, University of California, Los Angeles, CA 90024)

In this study, a search is made for high-noise-source regions for a jet impinging on a large flat plate. By using cross-correlation techniques for the pressure fluctuation on the plate and the farfield acoustic radiation, the location of the noise source using time delays (phase shifts) can be determined. Different jet speeds (up to Mach number 0.4), jet-to-plate distances (5 to 10 jet diameters), and impinging angles (0° to 80° from the jet axis) were used in an experiment in the UCLA Aeroacoustics Laboratory anechoic room. A full digitization treatment was used in analysis. The results show that the source is located on or near the plate in a circular region centered on the jet center, with a radius of approximately one jet diameter. The directivity pattern of the radiation will also be discussed. Standard theory indicates that there should be little surface, dipole sound from a large flat plate. The question will be discussed.

4:57

LL12. Structural resonance influences on flow-induced acoustic amplification. Alison Flatau (Mechanical Engineering, University of Utah, Salt Lake City, UT 84112)

An experimental investigation of the influence of structural resonance on the acoustic response to vortex shedding is presented. Flow-induced amplification of acoustic energy is of interest in the field of combustion instability in solid rocket motors. Baffles located at the junction of motor segments can cause generation of flow vortices and amplification of acoustic energy. This investigation examines the hypothesis that an additional amplifier of acoustic energy is provided by the vibratory response of these baffles. Flow through a cylindrical cavity is interrupted by annular baffles that extend radially inward from the cylinder walls. Baffle configurations are controlled to induce various frequencies of vortex shedding. Acoustic pressure and baffle vibratory responses for various combinations of baffle resonance and vortex shedding frequency are recorded using a modal analysis technique. This provides the database used to quantify the influence of vortex shedding frequencies and structural resonance on acoustic pressure amplitudes. A theoretical model of the interactions of different baffle configurations with chamber acoustics is presented and shown to be in general agreement with the acoustic pressure amplitudes measured in the test section. [Work supported by Morton Thiokol, Inc.]

5:09

LL13. On sound radiated by inhomogeneities convected through a space-varying flow. Alan Powell (Department of Mechanical Engineering, University of Houston, Houston, TX 77204-4792)

It was pointed out in Powell [J. Acoust. Soc. Am. **36**, 1032 (1964)] that inhomogeneities in composition, entropy, or vorticity give rise to sound radiation when convected through a *steady* space-varying flow as in a nozzle or past a curved surface. The one-dimensional case was extended by Cuadra [J. Acoust. Soc. Am. **42**, 725–732 (1967)] and more completely by Ffowcs-Williams and Howe (1975), but the initial "scattering approach" presented appears to have been published only qualitatively, e.g., Powell [Noise Control Eng. J. **8**, 108–119 (1977)]; but note related developments by Crighton and Leppington (1971), and Morfey (1973). This is simply Rayleigh scattering with an axis change so that the inhomogeneities sense time-varying changes as they are swept supersonically through weak oblique standing waves springing from a fixed sinusoidal wall. The physical arguments carry over for evanescent waves (complex wavenumber), the corresponding steady flow being subsonic. It is readily inferred, e.g., that for subsonic and supersonic flows (away from $M \simeq 1$), the sound power depends on the sixth power of the velocity.

5:21

LL14. Turbulent boundary layer simulation setup on a sonar dome. Jean Audet, Michel Lagier (Thomson Sintra ASM, route des Dolines, BP 38, 06561 Valbonne, France), Pierre Marin-Curtoud, and Thierry Rohan (Centre d'Etudes et de Recherches pour la Discrétion Acoustique des Navires, DCAN Toulon, 83800 Toulon Naval, France)

An experimental–analytical technique was developed to evaluate the noise produced inside various types of sonar domes under the effect of a turbulent flow, in a frequency range where the structure has a high modal density. The transfer functions between a punctual external force applied to the dome and the acoustic pressure at a point inside the cavity was measured on a scale model. This measure was made using a reciprocity technique with emitting hydrophones within the cavity and with accelerometers all over the dome's external surface. The noise level at a point inside the cavity is calculated from these experimental transfer functions and a turbulent boundary layer wall pressure model. The wave vector frequency spectrum model used is Chase's model [D. M. Chase, J. Sound Vib. **70**, 29–67 (1980)], adjusted according to the local parameters of the flow. The results obtained with this method agree with experimental results. In order to test various types of structure, this technique on a scale model has considerable advantages compared with experimentations on a real structure: limited expense, experimental ease, and reliability.

5:33

LL15. Wall pressure fluctuations beneath a transitional boundary layer. J. Audet and Ph. Dufourcq (DSA—Thomson–Sintra ASM, 525, route des Dolines Parc de Sophia Antipolis, BP 38-06561 Valbonne Cedex, France)

Up to now, different models have been developed to describe wall pressure fluctuations beneath a fully developed turbulent boundary layer [G. M. Corcos, J. Acoust. Soc. Am. **35**, 192 (1963); D. M. Chase, J. Sound Vib. **70**, (1)]. Recently, the total field pressure in the transition zone was described in different papers [G. C. Lauchle, J. Acoust. Soc. Am. **67**, 158–168 (1980); M. Lagier and D. Sornette, Acustica **61**, 116–124]. These works show that intermittent processes induce monopole sources whose radiation could be dominant. The present paper is devoted to the development and numerical analysis of a formulation based on these new models and fitted to wall pressure fluctuations calculation (local field). In this case, the wall pressure fluctuation is the sum of two terms: one associated with the turbulent content of intermittent spots and the other associated with the spot boundaries (monopole radiation). The relative contribution of each kind of sources to the transitional boundary layer wall pressure fluctuation can be investigated. Successful comparisons wth experimental results [Ph. Dufourcq, thèse Docteur Ingénieur, Ecole Centrale de Lyon (1984)] prove, in many cases, the dominance of monopole sources induced by intermittency. Noise generated by the transition region could be dominant at low frequency and could even disturb the upstream laminar flow.

5:45

LL16. Radiated sound-pressure levels and sound source locations on the German ICE high-speed train. E. Pfizenmaier, W. F. King, III, H. Lettmann,[a] and B. Barsikow (DFVLR, Turbulence Department, Mueller–Breslau-Strasse 8, D-1000, Berlin 12, Federal Republic of Germany)

The German ICE high-speed train demonstrated on 1 May 1988 that speeds up to 406 km/h can be attained using conventional wheel/rail technology. High-speed railway vehicles such as the ICE generate strong aerodynamic sound sources that overwhelm the noise contribution produced by wheel/rail interaction, particularly when the wheels have been acoustically treated. If radiated noise levels are to be held to a minimum, the aerodynamic characteristics of such vehicles have to be optimized. The ICE has been designed in this way. Sound-source location measurements were carried out using a line array of 15 microphones positioned along the wayside. The first results of this study will be discussed in the present paper for train speeds up to 300 km/h. Wayside noise measurements made during the record-breaking run on 1 May 1988, when a speed of 406.9 km/h was attained, are also presented. The aerodynamic component of radiated noise is separated from that due to wheel/rail interactions, and the resulting analysis is shown to be also applicable to maglev vehicles. [a] Also at Technische Universität Berlin, Inst. für Techn. Akustik.

WEDNESDAY AFTERNOON, 16 NOVEMBER 1988 MAUI ROOM, 2:00 TO 6:00 P.M.

Session MM. Underwater Acoustics V: High-Frequency Underwater Acoustics (Lecture and Précis-Poster Session)

Please note: The first six papers in this session will be presented in lecture format. Papers MM7 through MM21 will be presented in précis-poster format. Posters should be set up before 2:00 p.m. Following presentation of précis, all posters will be displayed until 6:00 p.m.

Steve Stanic, Cochairman
Naval Ocean Research and Development Center
Stennis Space Center, Mississippi 39529

Sumio Takahashi, Cochairman
Department of Applied Physics
National Defense Academy
1-10-20 Hashirimizu
Yokosuka, 239 Japan

Chairman's Introduction—2:00

Invited Papers

2:05

MM1. Modeling high-frequency bottom backscattering. Darrell R. Jackson (Applied Physics Laboratory, College of Ocean and Fishery Sciences, University of Washington, Seattle, WA 98195)

The recent acquisition of acoustic and geoacoustic data sets relevant to high-frequency bottom backscattering has made it possible to rigorously test models based on the composite roughness and Kirchhoff approximations. Examples will be given to show that these single scattering models have performed well within their expected regimes of validity and have helped improve our understanding of the relative importance of scattering from interface roughness and volume inhomogeneities. Some of the data show, however, that there are bottom types that do not fit within the idealizations of the simpler models. Single scattering approximations become increasingly suspect at small grazing angles, for very rough surfaces, and for volume scattering in soft sediments containing numerous buried scatterers. Improved models should incorporate multiple scattering, layering, and the fractal nature of the interface. [Work supported by ONT and NORDA.]

2:25

MM2. Acoustic transmission across the water/sediment interface. L. J. Satkowiak and K. L. Williams (Naval Coastal Systems Center, Code 2120, Panama City, FL 32407-5000)

Two sets of measurements were made to investigate the effect of the water/sediment interface on a propagating acoustic signal. In the first experiment, at-sea measurements utilized a three-dimensional array of hydrophones buried in the sediment as acoustic receivers. A linear acoustic source was mounted on a moveable tower located on the seafloor. The sound-pressure levels were measured as a function of grazing angle, frequency, and pulse shape for two distinctly different sediment types, fine sand and soft mud. The second set of measurements was made under nearly laboratory conditions. Data were collected mapping out the sound-pressure field in a homogeneous sand for both linear and parametric sources operating at a frequency of 20 kHz. The data were taken at grazing angles that ranged from well-above to well-below the critical angle for the sediment. Based on the data collected, in-sediment pressure field contours were obtained detailing the structure of the pressure field in the sediment. Both sets of data were compared with theoretical predictions calculated using the SAFARI model [H. Schmidt and F. B. Jensen, J. Acoust. Soc. Am. 77, 813–825 (1985)]. A discussion of the applicability of the SAFARI model will be presented.

2:45

MM3. High-frequency acoustics bottom interaction. Hollis Boehme and Nicholas P. Chotiros (Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78713-8029)

Over the past half century, a large amount of bottom interaction data has been collected by the scientific community. From a comparison of the extant high-frequency backscattering strength data, it is clear that the current schemes of bottom type classification, based on textural descriptions of the geology, make poor indicators of scattering strength. Recently obtained results suggest that bottom topography is as important as composition. The topography of the ocean bottom, in turn, is partly determined by oceanographic processes, such as currents and tides. Thus a new set of descriptors is evolving that will give better indications of bottom backscattering strength. Topography also exerts a strong influence on acoustic transmission through the interface. The effects include scattering and deformation of the wave front.

3:05

MM4. A review of high-frequency acoustic boundary scattering prior to 1982. Robert W. Farwell (Naval Ocean Research and Development Activity, Stennis Space Center, MS 39529-5004)

High-frequency acoustic sea surface and ocean floor scattering have attracted much attention from underwater acousticians over the past 50 years. In keeping with evolving system needs, the level of knowledge required from scattered signal characteristics has expanded. Many of these issues are currently being addressed. This paper will review the status of high-frequency acoustic boundary scattering measurements and models above 10 kHz. Both forward and backscatter strengths, coherences, higher-order statistical characterizations, as well as supporting environmental measurements, will be presented.

3:25

MM5. High-frequency sea surface scattering measurements. W. I. Roderick and J. B. Chester (Naval Underwater Systems Center, New London, CT 06320)

Measurement and modeling of forward and backscatter from the sea surface are important areas of study in high-frequency acoustics. To validate acoustic scattering models that involve reradiation from the surface and

near-surface volume scatterers (e.g., bubbles, biologics), acoustic measurement techniques are required that isolate the scattering mechanisms from multipath effects. In this paper, results from surface scattering measurements that obtained reverberation as a function of grazing angle utilizing both parametric and conventional projectors are presented. Measurements were conducted in shallow waters and the directivity properties of the parametric array isolated the surface scattered path. In the study of surface forward scatter, a new technique has been developed that allows separation of surface scatter and direct path by forming a vertical virtual aperture. This technique may be useful in situations where time resolution (bandwidth) is not sufficient. In terms of reverberation, surface scattering strength measured as a function of grazing angle (in 10-deg increments) will be presented for two different sea state conditions. Also, low grazing angle Doppler shift and spread will be shown as a function of up-wind and cross-wind conditions. These results were obtained in measurements conducted in the shallow waters east of Jacksonville, Florida.

3:45

MM6. High-frequency sea surface scattering: Recent progress. Suzanne T. McDaniel (Applied Research Laboratory, The Pennsylvania State University, University Park, PA 16804)

Recent progress in understanding the processes governing high-frequency acoustic scattering from the sea surface is reviewed. Theories for scattering due to surface roughness and from resonant microbubbles are presented, and the status of environmental inputs to these theories, including ocean roughness spectra and independently measured bubble populations, is discussed. Recently acquired acoustic data are shown to clearly demonstrate that resonant scattering from subsurface bubbles constitutes an important contribution for backscattering geometries at frequencies as low as 3 kHz. The theoretical predictions are then extended to lower frequencies, where resonant bubbles are not a viable scattering mechanism, and compared with backscatter data acquired using explosive charges. Whereas good agreement is obtained with certain data sets, other data lie consistently above the predictions. [Research supported by ONT with NORDA management.]

Contributed Papers

4:05

MM7. Backscattering and forward loss from near-surface bubbles. Steven O. McConnell (Applied Physics Laboratory, College of Ocean and Fishery Sciences, University of Washington, Seattle, WA 98195)

Measurements have been made of high-frequency (15–45 kHz) vertical incidence backscatter from near-surface bubbles generated by wave breaking. Interspersed with these backscatter measurements were measurements of the forward loss at low grazing angles owing to absorption by this near-surface bubble layer. In agreement with backscatter measurements from the same experiment reported earlier as well as those reported by others [see articles in *Noise in the Sea*, edited by B. Kerman (Reidel, New York, 1988)], the bubble volume scattering cross section decreases rapidly away from the surface usually displaying an exponential behavior with depth. Both the backscatter and forward loss data were taken using a rapid sequence of short pulses having different frequencies (100 ms between pulses). Thus far, the results show that the depth-integrated volume scattering strength averaged over a series of pings (typically 50–100 pings) increases linearly with frequency over the measured frequency interval. On a ping-to-ping basis, however, the fluctuations in this depth-integrated quantity show surprisingly little correlation between frequencies and even longer term temporal trends that appear at one frequency are not necessarily reproduced at other frequencies. [Work supported by NORDA.]

4:09

MM8. Environmental dependence of high-frequency volume scattering. D. V. Holliday (Tracor Applied Sciences, 9150 Chesapeake Drive, San Diego, CA 92123) and R. E. Pieper (University of Southern California, Institute of Marine and Coastal Studies, 820 South Seaside Avenue, Terminal Island, CA 90731)

Acoustic volume scattering strengths were measured in the upper 100 m at 21 frequencies between 100 KHz and 10 MHz. The scattering strength and its variability are examined with frequency and depth in the context of the local physical and biological environments. The data for this study were collected in the coastal transition zone off northern California, at the front that separates the Gulf Stream from the continental slope water near Cape Hatteras, in the Gulf Stream and in the southern California bight. [Work supported by ONR and NSF.]

4:13

MM9. Backscattering measurements in the presence of bubbles. Bernd Nützel and Heinz Herwig (Forschungsanstalt der Bundeswehr für Wasserschall- und Geophysik, Klausdorfer Weg 2-24, 2300 Kiel 14, West Germany)

The results from a recent acoustic scattering experiment, which was conducted in the North Sea, are presented with supporting environmental information. The acoustic data were obtained utilizing a high-resolution (narrow beamwidth) pulsed parametric sonar transmitter and conventional receiver. The measurements were done in the frequency regime from 20 to 80 kHz at grazing angles of 30° and 90° (normal incidence). The results show that the contribution of bubble clouds to total backscattering from the near-surface layer is negligible at normal incidence. However, bubbles can acoustically screen the surface so that backscattered energy from the surface can no longer be used to compute backscattering strength because of increased attenuation within bubble clouds. At a grazing angle of 30°, the backscattered energy is significantly influenced by bubble clouds when breaking waves occur.

4:17

MM10. Breaking waves and bubble clouds: Acoustic observations of the upper ocean boundary layer. David M. Farmer and Svein Vagle (Institute of Ocean Sciences, P.O. Box 6000, Sidney, British Columbia V8L 4B2, Canada)

When surface waves break, they modify the acoustic environment of the upper ocean boundary layer in two important ways. The breaking events generate sound, and the injection of bubble clouds modifies the near-surface, sound-speed distribution. Individual breaking events contribute to a spatial and temporal pattern of sound in the ocean that can be used to infer their temporal and spatial distribution on the ocean surface. The bubble clouds are organized by the boundary layer turbulence. Reduced sound speed within the clouds can refract sound generated by the breaking events and even selectively trap it within the surface bubble layer. Modification of the sound generated by breaking events may therefore serve also as a probe of the bubble layer properties. Observations of these processes will be described. The measurements were obtained with a novel free drifting instrument that employs both active and passive acoustic sensors.

4:21

MM11. Underwater noise due to the interaction of single water drops with the air/water interface. Eui Jun Kim, Kwang Joon Park, and Suk Wang Yoon (Acoustics Research Laboratory, Department of Physics, Sung Kyun Kwan University, Suwon, Republic of Korea 440-746)

Underwater noise due to the interaction of single water drops with the air/water interface is experimentally studied. In an individual water drop experiment, the noise level is increased as the water drop becomes larger, and its impact velocity is raised. In a consecutive water drop experiment, it is observed that a critical falling rate of water drops exists for abrupt change in noise level. The noise due to water drops larger than 3.42 mm in diameter becomes abruptly quiet above their critical falling rate. However, the noise due to water drops smaller than 3.11 mm in diameter has peaks in the frequency range of 12–18 kHz above their critical falling rate. These experimental results confirm that the bubble induced by the drop impact is a more dominant noise maker in water than the impact of a water drop on the air/water interface.

4:25

MM12. The effect of monomolecular films on the underlying ocean ambient noise field. Jim Rohr, Ray Glass (Naval Ocean Systems Center, San Diego, CA 92152), and Brett D. Castile (ORINCON Corporation, 9363 Towne Centre Drive, San Diego, CA 92121)

A series of at-sea tests has established unequivocally that the presence of a monomolecular surface film results in a pronounced reduction of ambient noise beneath the film. This reduction was observed over the wide range of surface conditions associated with sea states 1–6. Possible mechanisms through which the film affects the underlying ambient noise field are explored. [Work supported by DARPA and ONT.]

4:29

MM13. Reduced ocean surface backscattering from under an artificial sea slick. Brett D. Castile (ORINCON Corporation, 9363 Towne Centre Drive, San Diego, CA 92121), Jim Rohr, and Ray Glass (Naval Ocean Systems Center, San Diego, CA 92152)

Experiments were conducted in which acoustic backscattering from under an artificial ocean surface slick was compared with that in the absence of a slick. Initial tests were made with an AN/AQS-13 helicopter-dipping sonar. Subsequent tests were made using transducer heads from a MK-46 and a MK-48 torpedo mounted on a remotely steerable device that was suspended below a ship. The latter system allowed control over the acoustic frequency and angle of incidence at the sea surface and limited the size of the scattering area. At steep grazing angles of 40 deg or more, scattering strength under the slick was only about 2 dB less than that outside of the slick at 30 kHz, 4 or 5 dB less at 20 kHz, and 8 dB less at 10 kHz. The apparent reduction in surface scattering strength disappeared at shallow grazing angles. [Work supported by Office of Naval Technology.]

4:33

MM14. Correlation estimates of high-frequency ocean bottom backscatter. S. Stanic, K. Briggs, P. Fleischer, R. I. Ray, and W. B. Sawyer (Naval Ocean Research and Development Activity, Stennis Space Center, MS 39529)

Two environmental acoustic scattering experiments have been conducted in shallow water off the coast of Florida. The ocean bottom ranged from a smooth homogeneous medium sand to one where the surface roughness was entirely due to a coarse surficial shell layer. The acoustic transmitting system used a pair of narrow-beam parametric sources operating at secondary frequencies from 20–180 kHz. The receiving system consisted of a 16-hydrophone two-dimensional spatial array with broadband capabilities to 250 kHz. Ping-to-ping correlation estimates are presented as a function of frequency, grazing angle, azimuthal angle, and environmental conditions. Cross-correlation estimates as a function of hydrophone separation, frequency, and environmental conditions are also presented.

4:37

MM15. Characteristics of the SeaMARC II phase data. Haruyoshi Matsumoto, A. N. Shor (Hawaii Institute of Geophysics, University of Hawaii, Honolulu, HI 96822), and J. G. Blackinton (Seafloor Surveys International, Honolulu, HI 96814)

SeaMARC II is an interferometric, shallow-tow, long-range side-scan sonar system that is capable of producing bathymetry as well as side-scan images of the bottom in real time. To produce the bathymetry, phase differences between the echoes received by two linear acoustic arrays are measured. The phase difference data, sampled in travel time, show a slightly skewed, near-Gaussian distribution. The ambient noise and signal interference from the sea surface seem to cause increase of the phase data spreading. For SeaMARC II, the following are true: (1) Both the surface reverberation and the indirect echo path via water surface cause significant phase dispersion throughout the echo return time; (2) ambient acoustic noise is the main limiting factor for the angular resolution in a deep-water survey; and (3) the phase sample distribution is near-Gaussian and skewed in time. The relationships of these factors to the induced spread of the phase data are discussed. The current scheme for producing bathymetry from the phase difference data is discussed in the light of signal interference and phase sample distribution. Based on the Gaussian and isotropic ambient noise assumption, a simple analytical equation for the SeaMARC II phase data spreading is derived as a function of sonar parameters including the bottom backscattering coefficient. [Work supported by ONR.]

4:41

MM16. Use of high-resolution roughness power spectra in predicting bottom backscattering. Kevin B. Briggs and S. Stanic (Naval Ocean Research and Development Activity, Stennis Space Center, MS 39529)

Seafloor relief has been measured at very high (<1-cm) resolution with underwater stereo photogrammetry in the course of environmental characterization for high-frequency acoustic experiments. Information concerning bottom microrelief for the purpose of validating backscatter models has been heretofore expressed as rms height roughness. The validity of using Gaussian statistical methods to describe bottom relief is examined with the Hinich bispectrum test [Brockett *et al.*, J. Acoust Soc. Am. **82**, 1386–1394 (1987)] for normality in continuous data. Because bottom backscatter models such as the composite roughness model require the slope of the two-dimensional roughness power spectrum as an input parameter, generation of the two-dimensional relief spectrum is preferable to utilization of the rms relief or estimation from the one-dimensional spectra. Two-dimensional roughness spectra generated from photographs of rippled and smooth bottoms are compared to one-dimensional roughness spectra produced from the same photographs. Model and data comparisons are made for bottom backscattering using both roughness spectra.

4:45

MM17. Observations of weak scattering during AATE. Terry E. Ewart and Steve Reynolds (Applied Physics Laboratory, University of Washington, Seattle, WA 98105)

Under ice measurements of acoustic phase and amplitude were made in the Beaufort Sea during April 1985 in the AIWEX acoustic transmission experiment (AATE). Pulses of 2, 4, 8, and 16 kHz were sent every 0.49152 s over a 6.4-km path between a moored transmitter and a depth cycling receiver array. The receiving aperture covered 153 m during down cycles that were repeated every 4.8 min for 12 days. The observations exhibit levels of intensity scintillation much weaker than those observed in the open ocean. At the lower frequencies, the scintillation levels are representative of statistics predicted by Born or Rytov theory. Measurements made of the internal wave field by AIWEX investigators exhibit fluctuations a factor of 10 to 20 below those observed in the open ocean. The observed travel time fluctuations and second and fourth moments of the field are consistent with predictions made using the observed environmental statistics. Tests comparing variations in time with the vertical variability have been made.

4:49

MM18. The sound of cracking sea ice. David M. Farmer and Yunbo Xie (Institute of Ocean Sciences, P.O. Box 6000, Sidney, British Columbia V8L 4B2, Canada)

When ice cracks, it radiates energy both into the surrounding ice sheet and directly into the water. A goal in Arctic acoustics is the recovery of information about the cracking process, and thus the ice rheology, from the detailed structure of the acoustic emission. Recent observations, using an array of broadband hydrophones in the Canadian Arctic, have provided new insights on the processes of crack formation. Although it is possible to detect contributions leaking from the ice waveguide, including Crary waves, which can be scattered by discontinuities in the ice, most of the detected acoustic signal travels directly from the source. The resulting signal is interpreted with a moving source model analogous to that used in earthquake mechanics, modulated by fine structure in the cracking process. The fine scale and baseband components of the signal are related to vertical and horizontal scales of the developing crack.

4:53

MM19. Laboratory and field measurements of acoustic scattering from sea ice. K. C. Jezek (U.S.A. CRREL, 72 Lyme Road, Hanover, NH 03755), T. K. Stanton (Woods Hole Oceanographic Institution, Woods Hole, MA 02543), A. J. Gow (U.S.A. CRREL, 72 Lyme Road, Hanover, NH 03755), and M. Lange (Alfred Wegener Institut für Polarforschung)

The morphology of sea ice growing thermodynamically in calm seas is complicated. The ice is characterized by temperature and salinity gradients that are driven by the prevailing oceanic and atmospheric conditions, the gradients, in turn, affecting physical properties such as permeability, porosity, liquid brine content, and the structure of the ice–water interface. That interface, in particular, exhibits a range of dynamic characteristics from submillimeter, pure ice dendrites that grow down into the liquid to smoother cellular structures that are believed to develop when either thermodynamic growth is slowed or when the salinity of the liquid is much reduced. The objective of the work presented here is to measure and understand the role changing ice morphology has on the acoustic properties of sea ice. To that end, measurements over the frequency band from 120 to 800 kHz have been made of the scattering properties of sea ice when insonified by a narrow beam at normal incidence. Observations made in an outdoor tank at CRREL and from the research vessel Polarstern, which operated in the Fram Straits during May 1988, reveal that the reflection coefficient can vary by at least a factor of 5 depending on the composition and structure of the ice near the ice/water boundary. The dendritic interface typical of growing sea ice plays a particularly important role resulting in improved acoustic coupling into the ice. As seasonal evolution changes the interface from dendritic to either cellular or ablating, the reflection coefficient rises to about 0.25. [Work supported by ONR and the Alfred Wegener Institute.]

4:57

MM20. High-frequency acoustic scattering from under-ice microscale surface roughness. Garner C. Bishop (Building 679, Code 8211, Naval Underwater Systems Center, Newport, RI 02841)

The under-ice surface, characteristic of the interior Arctic, may be partitioned into three more or less distinct roughness regimes. Large-scale under-ice surface roughness is produced by discrete large-scale relief features, e.g., ice keels, and may be parametrized by ice depth that varies from several meters to several tens of meters. Small-scale surface roughness is produced by ice blocks and/or ice rubble, and may be parametrized by the maximum linear dimension of the ice blocks and/or rubble that varies from several tens of centimeters to several meters. Microscale surface roughness is present on the various surfaces of the ice blocks and/or rubble and is produced on the upper ice surface by snow and/or erosion, on vertical surfaces by the fracture of ice, and at the sea water–ice boundary by ice dendrites formed when saline ice freezes, and may be parametrized by an rms roughness parameter that varies from several millimeters to several centimeters. The micro-scale surface roughness produced by ice dendrites has been modeled by an array of close-packed, infinite and parallel semicylinders whose cross section can be varied from circular to elliptical. A Fredholm integral equation of the second kind has been developed and solved numerically to calculate the pressure field scattered from the microscale surface roughness. The effects of various boundary conditions and geometric parameters on the scatter of a high-frequency (>2 kHz) plane wave have been determined. [Research supported by the Office of Naval Technology and Naval Underwater Systems Center IR/IED.]

5:01

MM21. Evaluation of an electronically focused multipulse imaging system. A. P. Miller, S. O. Harrold, and J. M. Reeves (Department of Electrical and Electronic Engineering, Portsmouth Polytechnic, Anglesea Road, Portsmouth PO1 3DJ, England)

The scan rate of electronically focused imaging systems is dependent on three factors: the time for the electronics to create the transmit transducer energizing pulses, the ultrasonic propagation time, and the time to produce an image from the received signal. This paper describes a method for utilizing the idle time between the transmitted pulse and its reception at the receiving array. The system operation is based on linear arrays of 32 elements for both transmit and receive. Short bursts of ultrasonic energy, several cycles in duration, are focused sequentially on each resolution cell of the scanned volume. The focusing on transmit is achieved by suitably delaying the energizing pulses to the individual transducers. Focusing on receive is achieved by limiting the reception period for each transducer to the time during which a received echo is expected from the resolution cell at the focal point. This allows several pulses to be transmitted before reception of the first echo, which considerably increases the scan rate. This paper presents the initial results of a prototype system and compares them with a computer simulation, enabling an evaluation of the system and its potential to be made.

Session NN. Speech Communication VIII: Special Focus Session IIA, Measurement and Modeling of Speech Articulations and Articulators (Poster Session)

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Chairman's Introduction—4:00

Invited Papers

Please note: Posters in this session NN will be on display also during Session TT, and may be left up overnight. All posters will be displayed from 4:00 to 6:00 p.m. To allow contributors the opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 4:00 to 5:00 p.m. and contributors of even-numbered papers will be at their posters from 5:00 to 6:00 p.m. Posters from this session will be on display also during Session TT in same room on Thursday from 8:00 a.m. to 12:00 noon.

NN1. Stability of vowel formants based on a simple acoustic tube model and a tongue model. Yuki Kakita and Kiyoshi Honda (Department of Electronics, Kanazawa Institute of Technology, Kanazawa-Minami, 921 Japan)

Mapping from articulatory parameters to acoustic parameters for vowel(s) was examined based on a simple acoustic tube model of the vocal tract and on a three-dimensional model of the tongue. In the acoustic tube model, two articulatory parameters, i.e., the location (L) and size (S) of the constriction, were selected. The second formant frequency showed an interesting folded pattern, as L varies, in the vicinity of the formant frequency for the vowel /i/. However, only using L and S, no other quantal characteristics in Stevens' sense [K. N. Stevens, *Human Communication, A Unified View*, edited by P. B. Denes and E. E. David, Jr. (McGraw-Hill, New York, 1972), pp. 51–66] were observed. On the other hand, the tongue model demonstrated stable formant frequencies for the vowel /i/ as reported in our previous paper [O. Fujumura and Y. Kakita, *Frontiers of Speech Communication Research*, edited by B. Lindblom and S. Ohman (Academic, London, 1979), pp. 17–24]. The stabilizing effect of muscle contraction on formant frequencies is further discussed.

NN2. Articulatory dynamics in long and short vowel pairs. Katherine S. Harris (Graduate School, City University of New York, New York, NY 10021 and Haskins Laboratories, 270 Crown Street, New Haven, CT 06511), Peter Alfonso (University of Connecticut, Storrs, CT 06268 and Haskins Laboratories, 270 Crown Street, New Haven, CT 06511), and Thomas Baer (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511)

In English, several vowels, such as /i/ and /ɪ/ may be said to form pairs. The members have been described as "tense" and "lax" because, while they are believed to be grossly similar in vocal tract topography, the lax member of the pair is shorter in acoustic duration and represents a less extreme tract deformation. Temporal and configurational aspects of "tense" and "lax" pairs were investigated by an x-ray microbeam technique using the facilities at the University of Tokyo. Tongue, lip, and jaw trajectories were examined for a single speaker, producing multiple utterances of the form /əpVp/. Preliminary analysis showed that the duration of tongue movement trajectories reflected acoustic duration differences, but configurational differences were more complex. [This research was supported, in part, by grants from NINCDS and the Voice Foundation. The collaboration of the staff of the Institute of Logopedics and Phoniatrics, University of Tokyo, is gratefully acknowledged.]

NN3. Use of the x-ray microbeam system for the study of articulatory dynamics. Robert D. Nadler and James H. Abbs (University of Wisconsin, Waisman Center, 1500 Highland Avenue, Madison, WI 53705-2280)

The new University of Wisconsin low-dosage x-ray microbeam system is capable of tracking the movements of multiple (up to ten) small (2–3 mm) gold pellets attached to speech articulators. A variety of speech and nonspeech research questions are currently being addressed by investigators using the microbeam system. These include studies of linguistic issues, acoustic and articulatory modeling, speech motor control, articulatory kinematics, mastication, and swallowing. The system also permits digitization of wideband speech acoustic and other physiological signals. These signals, obtained along with articulatory pellet tracking, include multichannel EMG, glottal impedance, differential oral and nasal airflow rate, air pressure, rib cage and abdominal movement, and external lip LED marker movements. A brief description of the x-ray system operating characteristics and performance specifications as well as overall facility data acquisition and analysis capabilities will be provided. Sample data will be presented in order to illustrate recent findings and highlight important practical considerations associated with running microbeam experimental protocols. [Support for this facility is provided by NINCDS (NS-16373).]

NN4. Vocal tract dimensions obtained from magnetic resonance images. T. Baer (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511), J. C. Gore (Department of Diagnostic Imaging, Yale School of Medicine, 333 Cedar Street, New Haven, CT 06501), L. C. Gracco, and P. W. Nye (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511)

Serial magnetic resonance images (MRIs) of the vocal tract were obtained from two speakers of English during steady-state productions of nine isolated vowels including the four point vowels. Vowels were sustained for about 2.5 min, interrupted only by inhalations. Maintenance of a stable configuration was aided by a head positioner and by constantly hearing the target vowel. Images were obtained in the transaxial, and also in the sagittal and coronal planes for the four point vowels. Images representing 0.5- to 0.8-cm slice thickness were obtained at 0.5-cm intervals. Computer-assisted tracings of the boundaries of the vocal tract were used to construct a three-dimensional digital model of the airway. Using methods from the Haskins articulatory synthesizer, vocal-tract area functions were derived and vowels were resynthesized. Analysis of the first three formants showed good agreement with the subjects' original productions. Data on the relationship between the sagittal dimensions and the cross-sectional areas at various positions in the vocal tract were also obtained and will be reported. [Work supported by NIH, the Klingstein Fund, General Electric Co., and the Yale School of Medicine.]

NN5. Controlling an articulatory model. Peter Ladefoged (Phonetics Laboratory, Linguistics Department, UCLA, Los Angeles, CA 90024-1543)

Nobody knows what the brain tells the vocal organs to do when producing speech. As a starting point, a computer model has been built in which there are the following independent articulatory parameters: height and backness of the body of the tongue, lip height, lip width, lip protrusion, jaw opening, tip of tongue curling, tip of tongue advancing, tongue root advancement, tongue bunching, and larynx raising. An alternative specification of the body of the tongue uses front raising and back raising gestures, each of which can be combined with the other in different degrees. There are also several options for the interdependent action of the parameters. Instead of being independently specified, the lower lip and the jaw can be determined by the gesture required for the tongue, or the tongue height and lower lip position can be determined by the jaw position. Lip protrusion can be considered to be a coordinative structure that also specifies lipwidth. Larynx height and tongue root advancement can be considered to co-vary. These and other possibilities will be demonstrated in a model of articulatory-acoustic relations in a (public-domain) program on a Macintosh computer.

NN6. Speech production models—Constraints and control strategies. Gunnar Fant, Qiguang Lin, and Pierre Badin (Department of Speech Communication and Music Acoustics, KTH Box 70014, Stockholm, S-10044 Sweden)

Our modeling includes five areas of speech production. (1) Extension of the three-parameter vocal tract area function model of Fant (1960) to include bandwidths and to allow for variations of length and degree of asymmetry of the tongue section, and covariation of these parameters as well as overall VT length and configurations in the laryngeal region. (2) Algorithms for inverse transform, i.e., deriving VT parameters from formant frequencies. (3) Control strategies for synthesis of combinations of vocalic sounds from VT area function parameters. (4) A study of essential cavity structures of fricatives and stops with a discussion of source functions, in part based on data from Fant (1960). (5) Discussion of how source–filter interaction may be implemented and how it affects control strategies in formant synthesis based on underlying vocal tract simulations. [This work has been supported by grants from the Swedish Board of Technical Development and the Swedish Telecom.]

NN7. Automatic optimization of articulatory text-to-speech. C. H. Coker and S. Parthasarathy (AT&T Bell Laboratories, Murray Hill, NJ 07974)

Internal parameters of a text synthesizer are manipulated by an optimization strategy to obtain best fits to natural speech. Working units of the optimization are vectors of parameters for an articulatory model (usually

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one vector per phoneme) and vectors of time and speed of transition for each parameter. Criteria for optimization are evaluated over a time window spanning a target segment and its adjacent transitions. Problems with excess degrees of freedom ("ventriloquist effects") are minimized by the long evaluation window and by designating only certain critical parameters as eligible for adjustment for each phoneme. The most serious caveat is that certain sounds inherently produce greater error than others (e.g., fricatives because of variability in spectra of noise), and thus can overshadow subtle effects in adjacent sounds. Error criteria must be carefully chosen and weighted across time to focus attention on perceptually relevant points. Results confirm that synthesis with phoneme-like units can produce speech comparable to reasonably good frame-by-frame speech coders. A single prototype "interpolation function" is sufficient to generate parameter transitions, but length and time of transition must vary significantly with context, especially for the slow parameters that dominate vowels and semivowels.

NN8. Speech production by a vocal cords–vocal tract–vocal tract wall vibration model. Hisayoshi Suzuki and Takayoshi Nakai (Electronics Department, Faculty of Engineering, Shizuoka University, Hamamatsu, 432 Japan)

This research was aimed at finding the effects of finiteness in the mechanical impedance of the vocal organs on speech sounds. A speech synthesis program has been developed considering the vocal tract wall vibration that causes sound leakage from the vocal cavity to the nasal cavity even when the velum is closed. The radiation of sound from the outer surface of the vocal tract is not always negligibly small, and its effect is enhanced by the nasal cavity. The synthesis program has the parameters of the physical properties and pressure of ambient gases, as well as the mechanical impedance of a yielding vocal tract wall. The wall vibration is treated as a small perturbation in the area function of the vocal tract. The following items will be discussed in the paper: comparisons between vowels with and without vocal tract wall vibration, sounds from the mouth opening, nostrils, and vocal tract wall, and the distribution of the strength of the wall vibration along the vocal tract. The effect of the pressure of ambient gases on these items will also be discussed.

NN9. Measurements of articulatory dynamics using optoelectric and magnetic devices. Yorinobu Sonoda, Toshiharu Konomi, and Keisuke Mori (Department of Electronics, Kumamoto University, Kumamoto, 860 Japan)

This paper introduces two methods for measuring articulatory movements. Optical and magnetic sensing techniques were applied to recording tongue movements. The optical recording principle is based on the measurement of the phase shift of modulated light reflected from the tongue surface with less sensitivity to its reflection factor, and not on the measurement of the intensity of reflected light, which has been the usual practice. The sensor unit was composed of a small phototransistor and two LEDs, mounted together onto an artificial palate, and was attached to the hard palate. The prototype system consisted of three sensor units. The magnetic field sensing system was further developed to monitor tongue articulations. Design improvements were made on an earlier prototype system [Y. Sonoda, IEEE Trans. Magn. **10**, 954–957 (1974)]. The sensor unit was composed of a magnetic multivibrator bridge circuit with amorphous alloy cores. Using these sensing systems together with the previously reported PSD system [Y. Sonoda, J. Acoust. Soc. Am. **72**, 700–704 (1982)], experiments are now being conducted to observe simultaneously tongue, lip, and jaw movements.

Contributed Papers

NN10. Viscous flow analyses of glottal models using a finite element method. Hirohisa Iijima, Nobuhiro Miki, and Nobuo Nagai (Research Institute of Applied Electricity, Hokkaido University, N12, W6, Sapporo, 060 Japan)

Some numerical flow analyses for glottal models using a finite element method are shown. The basic equations to be analyzed are two-dimensional Navier–Stokes equations, where the airflow from the lung is assumed to be an incompressible viscous fluid. Both the steady-flow and the unsteady-flow analyses are performed. In the case of the steady-flow analysis, a Newton–Raphson iteration is used to obtain the solution. The generation and the growth of vortices and the complication pressure distributions were found to be affected by the viscosity. In the case of the unsteady flow, the velocity correction method is used to form the discrete finite-element equations. As a result, a dynamical downstream movement of the vortices was determined. Both in slow steady flow and fast unsteady flow vortices were generated in the glottis. These results indicate that the vortex at the glottis cannot be ignored for voiced sounds.

NN11. Pellet placement in x-ray microbeam studies. Mona Lindau and Peter Ladefoged (Phonetics Laboratory, University of California, Los Angeles, CA 90024-1543)

A weakness of the x-ray microbeam system for recording articulatory movements is the difficulty of placing pellets on the pharyngeal part of the tongue. It is, however, possible to use the system to obtain static position of the tongue-root in another way. Image scans of sustained English vowels of one subject, in which a chain of 15 pellets 5 mm apart had been placed on the midline of the tongue, were recorded and analyzed. The results enabled the determination of the best locations for placing pellets and the extent to which tongue shapes can be determined from a small number of pellets. If only two pellets are used, one should be placed in the range of 30–35 mm and the other 60–65 mm from the tip. The tongue shapes predicted from pellets in these locations were highly correlated ($r^2 = > 0.98$) with the observed shapes as specified by three factors that themselves determine the tongue shapes within 0.5 mm. The same high correlations ($r^2 = > 0.98$) can be achieved with several different combi-

nations of three pellets. Most (75%) of these combinations involve one point at the root of the tongue. However, by selecting points near the tongue tip (5–10 mm back), the middle (approximately 35 mm from the tip) and the back (45–50 mm from the tip), equally good correlations are obtained.

NN12. Vocal-tract areas versus articulatory parameters in speech production modeling. J. Schroeter, J. N. Larar, and S. Parthasarathy (AT&T Bell Laboratories, Murray Hill, NJ 07974)

Evaluation of articulatory codebooks [e.g., J. Schroeter *et al.*, J. Acoust. Soc. Am. Suppl. 1 **82**, S54 (1987)] has clearly shown that improvements can be realized by inserting codewords that reduce access error for certain problematic sounds (e.g., /r/, /l/). Originally, it was not clear whether the needed shapes are attainable with the articulatory model used to convert geometrical parameters of the tract model into area data. Comparing two simple articulatory models did not reveal major differences between them. Both seem to be able to cover the formant space adequately when driven by random parameter values that included formant configurations similar to those of /r/ and /l/. Different optimization algorithms, however, were only moderately successful in finding the global optimum in these cases. The approach taken to overcome this problem was to optimize vocal tract areas directly, thus eliminating any articulatory model. Instead of the geometric constraints inherent in an articulatory model, only loose constraints of specified bounds were imposed on each of the tract sections. These bounds were obtained from histograms of section areas computed from the largest articulatory codebook. In the analysis/synthesis system, area optimization showed superior performance over articulatory model parameter optimization.

NN13. Cross-language EPG data on lingual asymmetry. A. Marchal, E. Farnetani, W. J. Hardcastle, and A. Butcher (Institut de phonétique, UA 261 C.N.R.S., 29 Avenue R. Schuman, Aix-en Provence, France)

Conventional x-ray views of the vocal tract give only limited information about the lingua–palatal contacts occurring during speech production. From these types of data, it is not really possible to evaluate the real size and the exact shape of the vocal tract. When needed, the area at different sections must, in fact, be reconstructed using a mathematical model [S. Maeda, J. Acoust. Soc. Am. Suppl. 1 **65**, S22 (1979)]. A review of literature indicates that little attention is paid to the potential asymmetry of the articulatory gestures. This phenomenon appears, however, to be particularly crucial in speech pathology [Suzuki, Jpn. J. Oral Maxillo. Fac. Surg. **30**, 45–54 (1984)], where lingual and palatal asymmetry is often very severe and causes a poor speech intelligibility. But, even for normal speech, there is some evidence that lingual asymmetry may occur [Hamlet *et al.*, J. Acoust. Soc. Am. **79**, 1164–1169 (1986)]. Moreover, it has also been suggested that an asymmetrical tongue placement would be necessary to achieve the expected acoustic output for certain consonants [C. H. Shaddle, J. Acoust. Soc. Am. Suppl. 1 **77**, S85 (1985)]. To assess lingual asymmetry quantitatively, it is proposed that the following index be computed: $I_{as} = (N_a - N_b)/(N_a + N_b)$, where I_{as} is the index of asymmetry, N is the number of contacts, a is the right side, and b is the left side. The details of tongue to palate contacts for consonants and clusters in English, French, and Italian in nonsense syllables or words embedded in sentences were investigated. They were read by seven native normal speakers. The data demonstrate for all speakers an asymmetrical lingual pattern. Asymmetry is not language bound and seems to reflect a more general phenomenon. Alveolars appear to be more asymmetrical than velars. For stops, the asymmetry is most important at the articulatory release.

NN14. Tongue pellet data analysis by singular-value decomposition approach. Qiuzhen Xue, Robert Nadler, and James Abbs (University of Wisconsin, Waisman Center, 1500 Highland Avenue, Madison, WI 53705-2280)

In order to obtain tongue trajectory information during speech, three or four pellets attached to the tongue of subjects are tracked in real time by the University of Wisconsin x-ray microbeam system. For each pellet, two time series are produced that correspond to x and y pellet coordinates, respectively. These data are being used to build a model of vocal tract acoustics. The singular-value decomposition (SVD) method has been found to be a good way to approach the following questions (1) Are the data redundant? (2) Can we find a better way to represent the data set? (3) Can relationships among the different pellets be discerned? In this experiment, six data series are organized into a data matrix and decomposed by SVD into two orthogonal matrices and one diagonal matrix. The diagonal matrix, called the singular-value matrix, indicates the redundancy of the original data set. Different deductions of the singular-value matrix were tested for five data sets. It is found that three transformed orthogonal data vectors can represent the original six data series at a 98% level of approximation. This work indicates that the orthogonal data matrices are well suited for model construction because: (1) the data size is decreased; (2) ambiguity is reduced for mapping work (e.g., to the terminal acoustic characteristics); and (3) relationships among the pellets can be found from the orthogonal matrices. These results and some other related issues will be discussed. [Work supported by NIH Grant NS-16373.]

NN15. Three-dimensional representation of articulatory kinematics from x-ray microbeam data. John R. Westbury and Qiuzhen Xue (Waisman Center, University of Wisconsin, Madison, WI 53705-2280)

Records of speech-related movements of flesh points and anatomic landmarks obtained from the University of Wisconsin x-ray microbeam system are fundamentally two dimensional in the sense that they provide time histories of relative positions of points within a plane. However, the third spatial dimension associated with each point is implicit in its two-space representation and, under special conditions, can be reconstructed mathematically. For example, three points attached to a rigid body such as the mandible determine a system of three polynomial equations, which, when solved, provides an account of the location of each point with respect to sagittal *and* coronal plane axes. Both the feasibility and resolution limits of a conventional numerical approach to this problem have been established by analysis of calibrated movements of three-point arrays attached to a machine-driven phantom and tracked by the microbeam system. Moreover, the same analysis procedure has been applied to kinematic data for multipoint arrays attached to the head and mandible of nine adult subjects to determine whether and to what extent their respective movements, during speech and other selected oral motor behaviors, are restricted to the sagittal plane. The advantages of a three-dimensional representation of speech motor behavior for at least these structures, within the context of recent microbeam experiments, will be discussed. [Work supported by NIH Grant NS-16373.]

NN16. Analysis of tongue motion for the dental consonants based on the high-speed palatographic data. Takao Mizutani, Kiyoshi Hashimoto (University of Electro-Communications, 1-5-1 Chofugaoka, Chofu, 182 Japan), Masahiko Wakumoto (Showa University, 2-1-1 Kitasenzoku, Ohta-ku, Tokyo, 145 Japan), Hareo Hamada, and Tanetoshi Miura (Tokyo Denki University, 2-2 Kanda-Nishiki-cho, Chiyoda-ku, Tokyo, 101 Japan)

Using a new high-speed palatograph system, tongue motions for dental consonants were recorded at a rate of 250 frames per second, and contact patterns of different consonants were compared using the gross features of the contact area (front–back position and high–low position).

Each of five male subjects uttered 30 nonsense English syllables. /hə'CV/ (C = /t,d,n,s,z,r/, V = /a,i,u,e,o/). Then the front–back position of the contact area is represented by the position of the center of gravity of the contact area. Information on the vertical position of the tongue was expressed by the total number of contact points between the tongue and the palate. Representation of the contact pattern in the two-dimensional space of these parameters shows the characteristic feature for each utterance. A distinctive feature among the plosives /t,d,n/ is that the tongue position is more frontal and/or higher for /t/ than for the other two plosives /d,n/. As for the fricatives /s/ and /z/, the tongue position for the latter is usually more frontal than for the former. [Work supported by MESC of Japan.]

NN17. Applying the program NEWPAR to extract dynamic parameters from movement trajectories. Caroline L. Smith, Catherine P. Browman, and Richard S. McGowan (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511)

In a model of articulatory movement, based on a dynamical systems approach, parameter values derived from actual articulatory data are being used. One way of obtaining these values relies on the program NEWPAR, which has been developed to analyze articulatory movement in order to extract dynamic parameters such as frequency that will serve as coefficients in the model equation [McGowan *et al.*, J. Acoust. Soc. Am. Suppl. 1 **83**, S113 (1988)]. Alternative hypotheses for the application of NEWPAR have been tested, including starting the half-cycle windows at different points in the trajectory and fixing either the initial displacement and velocity or the initial and final displacement as a means of reducing the number of parameters to be fitted. Tests have been made on simulated data, consisting of sinusoidal curves with damping varied from completely undamped to critically damped. Patterns in the results of these tests and some preliminary analyses of actual lower lip movement trajectories will be shown. [Work supported by NSF and NIH.]

NN18. Effects of speaking rate on formant trajectories and their interspeaker variations. Satoshi Imaizumi and Shigeru Kiritani (RILP, Faculty of Medicine, University of Tokyo, 7-3-1 Hongo, Bunkyo-ku, Tokyo, 113 Japan)

Variations in formant trajectories due to speaking rate were investigated for /VCVCV/ utterances, V∈[a,i] and C∈[b,d,g], which were recorded from three normal speakers and two Broca's aphasics with apraxia of speech. For the normal subjects, the timing of the F_3 transition toward the following stop consonants relative to the onset of the vowels, and also the transition rate of F_3, showed large variations depending on the speaking rate. The magnitude of the timing and rate changes varied depending on the subjects. As for F_1 and F_2, variations in the onset frequencies and the transition rates due to the speaking rate were smaller than the interspeaker variations. For the aphasics, the formant trajectories showed large interutterance variations, but no systematic variations as a function of speaking rate. [Work supported by Grant-in-Aid for Scientific Research on Priority Areas, the Ministry of Education, Science and Culture, Japan.]

NN19. Vowels formants in different modes of speaking. Noriko Umeda and Karen Wallace (Institute for Speech and Language Sciences, New York University, 719 Broadway, New York, NY 10003)

Articulatory differences between maximally articulated situations and fluent speaking are said to lie in the degree of articulatory overlapping between neighboring phonemes, and target undershooting due to lack of execution time when the phoneme becomes short. In addition to these segmental factors, however, global factors may also cause a certain mode of speaking distinctively different articulatorily from other situations. For example, more limited jaw movements in fluent speaking may result in smaller F_1 movements than words in isolation. The effect of corner cutting while concentrating efforts on selected items may be at work in fluent speaking. In order to capture comprehensive articulatory strategies, a study on formant frequencies of vowels in various modes of speaking has begun. A preliminary result with one speaker shows that, at the onset of vowels from consonants in the stressed syllables of content words, F_1, representing roughly jaw movements, varies far greater a range in carrier phrases than in fluent situations; F_2, which represents roughly back-and-forth movements of the tongue varies least in sentence-reading and greatest in conversation.

NN20. Aerodynamics in two-dimensional vocal tract models. James DeLucia (IDA/CRD, Thanet Road, Princeton, NJ 08540)

A numerically tractable formulation of the Navier–Stokes equations, suitable for studying fluid flow in the vocal tract during speech production and for doing more natural speech synthesis, is described. These qualities are obtained by restricting consideration to two-dimensional vocal tract geometries and by introducing a velocity stream function $A(\psi,\theta)$ and a velocity potential $\Omega(\psi,\theta)$. Here, (ψ,θ,ϕ) are curvilinear coordinates with ϕ being ignorable. Taking the divergence and curl of the momentum equation isolates the compressible wave motion in the velocity potential and the background "quasi-steady-state" flow in the stream function. This analytic separation of the two flow components allows for dramatic improvements in the accuracy and efficiency of any numerical simulation as compared to those obtained by directly integrating the primative equations. Full steady-state flow is represented compactly by a "momentum density stream function" $\Phi(\psi,\theta)$. For invisid flow, a second-order elliptic PDE for Φ is obtained with the fluid density given self-consistently by Bernoulli's law. With viscosity, a higher-order equation for Φ results. Details of the formulation, geometry, and boundary conditions are presented

NN21. Kinematic analysis of articulatory declination. Eric Vatikiotis-Bateson (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511) and Carol A. Fowler (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511 and Dartmouth College, Hanover, NH 03755)

Previous investigations of declination—the progressive reduction of some measure over the course of an utterance—have focused primarily on $F0$, with occasional mention of acoustic amplitude. Recently, $F0$ declination has been shown to track a decline in subglottal pressure. To our knowledge, whether declination occurs in other subsystems associated with speech production has not been investigated. One such subsystem currently being examined for speakers of English, French, and Japanese is that of supralaryngeal articulator motion. Kinematic variables of displacement, duration, and peak velocity, associated with motion of the lower lip/jaw complex during reiterant speech, were tested for declination effects. Initial findings show that there is articulatory declination for some speakers, but there is much interspeaker variability when kinematic variables are examined individually. However, as demonstrated previously for stress and speaking rate contrasts, much of the spatiotemporal variability may be accounted for when the relations among kinematic variables are considered, thereby making it easier to determine whether articulatory declination is present. Furthermore, for the data examined thus far, articulator motion appears to adhere to the constraints of a damped, second-order dynamical system, whether articulatory declination is observed or not. [Work supported by NIH.]

NN22. Some factors affecting vowel articulation. Alice Faber (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511) and Lawrence J. Raphael (Herbert H. Lehman College, Bronx, NY 10468, the Graduate School, City University of New York, New York, NY 10021, and Haskins Laboratories, 270 Crown Street, New Haven, CT 06511)

As part of an extensive series of studies of tongue activity in vowel production and of the muscle activation patterns producing this activity, data were gathered from one speaker (LJR). Four sets of 15 repetitions of 15 English vowels and diphthongs in a /pVp/ context were produced, in four separate conditions ($\pm$ bite-block, rested versus fatigued). Acoustic analysis of each token was undertaken, along with analysis of the electromyographic signals, recorded, in parallel, from seven intrinsic tongue muscles and the lower *obicularis oris*. The vowels in the bite-block conditions differed from the corresponding non-bite-block vowels in $F1/F2$ values and in the consistently stronger EMG signals recorded for the former. In addition, tokens from the "fatigued" conditions were shorter than those from the "rested" conditions and, for some vowels were characterized by a markedly higher $F0$. [Work supported by NIH.]

NN23. Stable lingual gestures in vowel articulation. P. J. Alfonso (University of Connecticut, Storrs, CT 06268, and Haskins Laboratories, 270 Crown Street, New Haven, CT 06511) and S. Horiguchi (University of Tokyo, Tokyo, 113 Japan)

Lingual movements monitored by lateral cineradiographic pellet tracking along with first and second formant frequencies for the articulation of /i/ and /u/ in labial, alveolar, and velar stop (ɔCVC/ syllables were analyzed in order to quantify both the relatively stable and more variable components of vowel-related tongue movements that occur as a function of stop coproduction. Horizontal and vertical tongue displacements varied by as much as 10 mm at tongue body locations for /i/ and 6 mm at tongue blade and anterior tongue body locations for /u/. Formant frequencies varied correspondingly. However, vertical displacement of the tongue blade for /i/ and of the posterior tongue body for /u/ were much more stable. These relatively stable displacement components suggest that the characteristic gestures for /i/ and /u/ act to create critical oral cavity apertures by tongue blade and posterior tongue body vertical displacements, respectively. The precise anterior–posterior locations of the lingual constriction are not specified in the gestures and are free to vary with consonantal coproduction. [Work supported by NIH NS-13617 and NS-13870.]

NN24. The effect of wearing aerodynamic measuring devices on speech articulation. James J. Mahshie (Department of Audiology and Speech, Gallaudet University, Washington, DC 20002)

Aerodynamic measures of speech production have been used to obtain insights into the dynamic properties of the vocal tract during speech production. It is unclear, however, how wearing devices such as the pneumotachograph and oral pressure transducers affects the articulatory processes being studied. The present research examined the articulatory characteristic of speech production by two normal speakers under two conditions: (1) while wearing a Rothenberg type pneumotachograph mask and oral pressure transducer and (2) without these devices. The x-ray microbeam data were obtained for lip, jaw, and tongue movement during multiple productions of voiced and unvoiced plosives in varied vowel contexts. Preliminary analysis suggests that articulatory dynamics appear not to change as a result of wearing the mask, although movement amplitudes may. Comparisons between the two conditions will be presented, together with implications for use of such measures as indices of speech articulatory behavior.

WEDNESDAY AFTERNOON, 16 NOVEMBER 1988 OAHU/WAIALUA ROOM, 4:25 TO 6:00 P.M.

Session OO. Architectural Acoustics VI: Perceptual and Physical Assessment of Auditoriums

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Contributed Papers

4:25

OO1. Preference test of seven European concert halls. Masayuki Morimoto, Zyun-iti Maekawa (Faculty of Engineering, Kobe University, Kobe, 657 Japan), Hideki Tachibana (Institute of Industrial Science, University of Tokyo, Tokyo, 106 Japan), Yoshio Yamasaki (School of Science and Engineering, Waseda University, Tokyo, 160 Japan), Yoshio Hirasawa (Acoustic Research Laboratory, ONKYO Co., Neyagawa, 527 Japan), and Christoph Pösselt (Ruhr University, D-4630 Bochum, Federal Republic of Germany)

Preference tests of seven European concert halls using musical motifs recorded through a dummy head were performed. Test signals were reproduced binaurally. Eighty-eight students served as subjects. The results show that (1) there was no significant difference between preference scores for these halls when averaged over all subjects, and (2) subjects could be divided into several groups with respect to preference. In conclusion, even though each subject was free to prefer hall A to hall B, the acoustical quality of a certain hall could not be solely evaluated using the preference score. [Work supported, in part, by The Kajima Foundation.]

4:37

OO2. Evaluation of the sound field in a room using PICC. Kiminori Yamaguchi, Takahiko Ono (Ono Sokki Co., Ltd., 2-4-1 Nishi-shinjuku, Shinjuku-ku, Tokyo, 163 Japan), Hirokazu Mizota, and Yasuhiko Yahata (Sound Craft Inc., 75 Waseda-cho, Shinjuku-ku, Tokyo, 162 Japan)

To find an adequate physical parameter that can explain the subjective impression of the sound field properties of a room, the PICC (perceptual interaural cross-correlation coefficient) is discussed. The PICC is a special interaural cross-correlation coefficient taking into account Haas' effect on hearing characteristics. Two experiments are reported in this paper. The first experiment was done in an actual auditorium with a volume of 9000 m^3. The speech signal sources were recorded at several seats in this auditorium; then a listening test on the subjective evaluation of the spatial impression was performed using these signals. At the same time, the PICC was measured by the synchronous response-multiplication method at each seat. The experimental results show agreement between the psychological scale of the spatial impression and the value of the PICC. The second experiment examined whether the PICC is useful in explaining the size of the sound image of reproduced sound sources in a small room. Here, a listening room with a volume of 50 m^3 was used. Several sound sources were prepared using comb filters. The results of the listening test on the size of the sound image were correlated with the value of measured PICC for each of the sound sources.

4:49

OO3. A study on the subjective effects of an active-control system for sound fields in an auditorium. Yoichiro Kato (SFC Inc., 2-3-4 Funahara, Itami, 664 Japan and Faculty of Engineering, Kobe University, Rokkodai, Nada-ku, Kobe, 657 Japan) and Yoichi Ando (Faculty of Engineering, Kobe University, Rokkodai, Nada-ku, Kobe, 657 Japan)

This paper presents subjective effects of an active-digital-control system for sound fields in an auditorium. A pilot study using various kinds of music was made to optimize a multipurpose hall in actual concerts. A wide range of music from baroque to contemporary, such as chorus, flute sonata, string quartet, wind quintet, and music for string ensemble was performed on the stage of a hall with 915 seats. In order to control early reflections and reverberation time as well as the IACC, signals were picked up by several microphones close to the sound sources. These signals were then fed to eight loudspeakers located at the ceiling above the seats after passing through digital-delay machines and digital reverberators. Questionnaires asking subjective preference, loudness, clearness, reverberance, diffuseness, and other related items such as experiences of attending concerts, performing music, etc., were distributed to about 200 listeners. The data for both conditions, on and off, of the system were analyzed. The results indicate that subjective preference for sound fields may be improved by a carefully adjusted active-control system for a wide range of seating areas.

5:01

OO4. Geometric and dynamic effects of stages in concert halls. James B. Lee (Concert Acoustics, Box 18017, Portland, OR 97218)

Wallace Clement Sabine's place in the history of concert halls is definitive, so much so that many have forgotten the seminal work of Joseph Henry, even though they use it implicitly in their practice. But Henry, writing from 1856, still has much to tell us: His famous echo experiment quantified the boundary between perception in the time domain and the frequency domain in terms of architectural scale—about 8 m—which implicitly has been made the boundary between large and small rooms. Concert halls are rightly thought of as large rooms. But look at the stage of Boston Symphony Hall: 8 m deep, bounded by an open trapezoid of very hard and very plane walls! If it were smaller, an orchestra would not fit; if it were larger, players would hear echoes and ensemble would suffer. This "Henry" scale stage works in the frequency domain too: Bass viols, with their low E of 41 Hz, $\lambda = 8$ m, are less than 1/4 wavelength from a wall; not only is their sound reflected geometrically, but also its source strength is augmented due to the enhanced impedance of radiating in a pressure zone.

5:13

OO5. Acoustical design of Tokyo Geijutsu Bunka-Kaikan. Part 2. Room acoustical design of the large hall. Minoru Nagata, Satoru Ikeda, and Keiji Oguchi (M. Nagata Acoustic Engineer & Associates Company, Ltd., 10 Shinano-machi, Shinjuku-ku, Tokyo, 160 Japan)

The large hall of the Tokyo Geijutsu Bunka-Kaikan is planned as a concert hall for classical music with 1887 seats. At the first stage of the design, the room shape and seating arrangements proposed by the architect were studied by computer simulation, in which the energy distribution of the early reflections over the seating area was observed. The room shape was repeatedly studied and re-designed considering the architectural design policy and acoustical characteristics. At the final stage of the design, a ray experiment in a 1/50 scale model was introduced. Acoustical experiments in a 1/10 scale model are planned in order to predict the room acoustical parameters and to check on detrimental echoes. The room acoustical design policy and the details of the new auditorium will be presented.

5:25

OO6. A study of the characteristics of early reflections in concert halls. Yasuhisa Toyota, Keiji Oguchi, and Minoru Nagata (M. Nagata Acoustic Engineer & Associates Company, Ltd., 10 Shinano-machi, Shinjuku-ku, Tokyo, 160 Japan)

Among many room acoustical parameters except RT60, clarity and room response (the C–RR chart) have been adopted and measured in several halls. But C is relatively dependent on reverberation time and the evaluation by the C–RR chart is not always satisfactory enough to explain the acoustical features of concert halls. As a new method for evaluating concert hall acoustics, "RECC" (reflection energy cumulative curve excepting direct sound) has been studied. The results are as follows. (1) RECC, up to about 80 ms, shows various shapes depending on the hall and measuring points. But, after about 80 ms, it shows a smoothly increasing convex shape for all halls. (2) Total reflection energy is almost determined by the ARE80 (accumulated reflection energy up to 80 ms). (3) The values of the ARE80 level are comparatively large at seats where one can feel intimacy in the hall. Room acoustics of concert halls is evidently very dependent on its room shape. The ARE80 is considerably influenced by room shape and independent of RT60. Optimum level, density, direction, frequency characteristics, etc., of the reflections up to 80 ms are problems for further study.

5:37–5:40
Break

5:40

Informal presentation and dissussion—"Recent Japanese efforts on concert hall acoustics," by Hideki Tachibama, Yoshio Yamasaki, and Masaru Koyasu, School of Science and Engineering, Waseda University, Tokyo, 160 Japan.

Session PP. Engineering Acoustics V: Signal Processing in Difficult Environments

Harry B. Miller, Cochairman
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Chairman's Introduction—8:00

Invited Papers

8:05

PP1. Real-time 3-D ultrasonic diagnostic imaging. Olaf T. von Ramm (Department of Biomedical Engineering, Duke University, Durham, NC 27706) and Stephen W. Smith (CDRH,FDA, Rockville, MD 20857)

Real-time phased-array two-dimensional ultrasound imaging has gained worldwide acceptance as a valuable diagnostic tool. In the past years, a parallel receive processing scheme for phased-array systems, called Explososcanning, which permits imaging of several B-mode lines for each transmitted acoustic burst, has been developed. Now this receive processing is being utilized in conjunction with newly developed two-dimensional arrays to electrically scan the ultrasound beam in three dimensions. The current 2-D transducers consist of 256 elements, operate at approximately 2 MHz and measure 16 by 16 mm at the site of skin contact. The 3-D echo data obtained from the pyramidal scan volume can be displayed as multiple sectors at preselected azimuth and elevation directions, as a two-dimensional projectional image with perspective or as stereoscopic pairs. Parallel processing will permit the formation of approximately 9000 look directions within the pyramidal scan volume at TV rates.

8:25

PP2. Statistical model-based speech enhancement systems. Yariv Ephraim (Speech Research Department, AT&T Bell Laboratories, 600 Mountain Avenue, Room 2C-572, Murray Hill, NJ 07974)

This paper deals with the problem of enhancing speech signals that have been degraded by statistically independent quasistationary noise. The estimation of the clean speech waveform, and of the parameters of autoregressive (AR) models for the clean speech, given the noisy speech, is considered. The two problems are demonstrated to be closely related in the sense that a good solution to one of them can be used for achieving a satisfactory solution for the other. The difficulties in solving these estimation problems are mainly due to the lack of explicit knowledge of the statistics of the clean speech signal and of the noise process. Maximum likelihood estimation solutions that are based upon the E–M algorithm and its derivatives are proposed. For estimating the speech waveform, the statistics of the clean speech signal and of the noise process are first estimated by training a pair of Gaussian AR hidden Markov models, one for the clean speech and the other for the noise, using long training sequences from the two sources. Then, the speech waveform is reestimated by applying the E–M algorithm to the estimated statistics. An approximation to the E–M algorithm is interpreted as being an iterative procedure in which Wiener filtering and AR modeling are alternatively applied. The different algorithms considered here will be compared and demonstrated.

8:45

PP3. An historical review: Two demonstrations of restoring old acoustic recordings. Thomas G. Stockham, Jr. (Computer Department, University of Utah, Salt Lake City, UT 84112)

This presentation demonstrates two complementary approaches used in the early 1970s for the experimental restoration of acoustic recordings of Enrico Caruso. The first approach employs a method of blind deconvolution that attempts to eliminate the reverberant and resonant characteristics of such recordings. The second approach attempts to remove the orchestral background using an analysis–synthesis method.

9:05

PP4. Study on a "spatially selective" microphone system. Hikaru Date (Department of Information Engineering, Yamagata University, Yonezawa, 992 Japan)

Spatial selectivity of a microphone system is defined as a function that rejects all sound signals coming from the region specified as external by the system. It can be effectively realized by signal processing of the multiport output signals of a receiving system that comprises several microphone pairs located on a virtual spherical boundary surface and one center microphone. Because the signal processing part of the system consists of adders, constant multipliers, differentiators, and a single delay, it is essentially linear and can be realized in either analog or digital form according to requirements of precision. The practical advantage of the spatially selective microphone system is that it has the capability of suppressing howling in public address systems and conference telephony, of improving the signal-to-noise ratio of microphone output to automatic speech recognition machines, and of improving the separability of the signal picked up from a particular musical instrument from those of other instruments played simultaneously in the same studio. The design method is presented after the simulation results for the system.

9:25–9:35
Break

9:35

PP5. Estimation of position and waveform of individual sound sources using many sensors. Ken'iti Kido, Yoshifumi Nagata, and Masato Abe (Research Center for Applied Information Sciences, Tohoku University, Katahira, Sendai, 980 Japan)

This paper describes a method for estimating the position and waveform of individual sound sources using many sensors when there are many sound sources. First, an imaginary sound source is introduced, and the distance between the imaginary sound source and every sensor is calculated. Next, the output of every sensor is compensated both in time and in magnitude so that the compensated output is the same for the sound radiated from the imaginary sound source. Then, the compensated outputs are averaged, and the power of the averaged waveform is calculated. When the position of the imaginary sound source just coincides with one of the positions of the real sound sources, the power has a large value. Otherwise, it has a small value. The position of the imaginary sound source is scanned within a specified space. The position, at which the power yields a peak, is the estimated location of a real sound source, and the averaged waveform is the estimated waveform of the sound source. Then, the perturbation of the estimated locations and waveforms is carried out. The final estimated values are those obtained when the error function reaches its smallest value. Computer simulations and experiments were carried out using two sound sources and 16 sensors, and good results were obtained.

9:55

PP6. Measurement of spatial information in sound fields by a closely located four-point microphone method. Yoshio Yamasaki and Takeshi Itow (School of Science and Technology, Waseda University, 3-4-1 Okubo Shinjuku-ku, Tokyo, 165 Japan)

When an estimation is made of the sound field in a room, it is important to grasp spatial information, especially for the early reverberation periods. Here, a way to grasp spatial information of sound fields from impulse responses measured at four closely located points, the origin, and three points the same distance ($3\sim5$ cm) from the origin on the rectangular coordinate axes will be discussed. From these four impulse responses the position of virtual image sources are calculated by the correlation and/or intensity technique. Concert halls in Europe, the United States, and Japan and some scale models are measured by this method. The three-dimensional distribution of virtual image sources, directivity patterns, and impulse responses from several directions are shown. Calculation of signals from special directions by convolving these impulse responses with dry music to estimate the sound fields with ears or to help the acoustic design, and the introduction of the Wigner distribution are also discussed.

Contributed Papers

10:15

PP7. Angular dependence of backscattered intensity of rough surfaces at different frequencies. K. Soetanto,[a] S. Ohtsuki, and M. Okujima (Tokyo Institute of Technology, Tokyo, Japan)

The relation between the scattered intensity and the roughness of a surface was measured by the angle of incidence at several frequencies with a typical pulse–echo system. Four types of phantoms with roughness, (a) smooth, (b) 0.03 mm, (c) 0.1 mm, and (d) 0.3 mm, are made of agar with one of the surfaces made rough for the observations. For these tests, 2.25-, 3.5-, 5.0-, and 7.5-MHz transducers are used. The angle of incidence [θ] ranges from 0° to 60°. The scattered intensity of the (a) and (b) phantoms was found to rapidly decrease by -25 to -38 dB as the angle increased. The decrease of scattered intensity with angle was less for phantoms with rougher surfaces. Smaller variation in scattered intensity at 7.5 MHz, compared to that at the low frequency of 2.25 MHz, is because of stronger scattering occurring from the rough surface when the wavelength is shorter. The rougher surface exhibits the smaller variation in scattered intensity with changing angle of incidence. The results of these measurements agree with earlier results [Reid, in *Diagnostic Ultrasound*, edited by C. C. Grossman *et al.* (Plenum, New York, 1966), pp. 1–12].[a] Presently at: Biomedical Engineering, Drexel University, Philadelphia, PA 19104.

10:27

PP8. Three-dimensional extraction and reproduction of signals from an interfering sound field. Kimitoshi Fukudome (Department of Acoustic Design, Kyushu Institute of Design, 4-9-1 Shiobaru, Minami-ku, Fukuoka, 815 Japan)

This paper describes a method for the three-dimensional reproduction of any constituent sound signals existing in the interfering sound field. The method is composed of three parts. (1) Estimation of incident directions and spectra of unknown sound sources using sphere-baffled microphones and the diffractive information of the sphere [K. Fukudome, J. Acoust. Soc. Jpn. **44**, 272–281 (1988)]. (2) Extraction of sound source signals by adding the output signals of the extracting filters that are fed by the sphere-baffled microphones. Characteristics of the filters are determined according to the estimated results of the incident directions and the diffractive information. (3) Three-dimensional reproduction of any constituent sound signals using headphones. The right and left signals of the headphones are generated after convolving the head-related transfer characteristics with the constituent sound source signal. The validity of the present method has been verified by computer simulation. Finally, the problems to be considered at the implementation are described. [Work supported partly by The Sound Technology Promotion Foundation.]

10:39

PP9. Dolphin sonar target detection in noise. Whitlow W. L. Au (Naval Ocean Systems Center, P. O. Box 997, Kailua, HI 96734)

Several different types of experiments have been conducted to determine the sonar detection capability of the Atlantic bottlenose dolphin. In one experiment, a dolphin's performance in detecting the presence or absence of a 7.62-cm-diam (-28.3-dB target strength), water-filled, stainless steel sphere as a function of range was determined. The threshold range in Kaneohe Bay, Oahu, Hawaii was approximately 113 m. In another experiment, the same sphere was fixed at a range of 20 m, and the detection capability of two dolphins was determined as a function of the level of artificial masking noise. In a third experiment, a simulated phantom electronic target was used to simulate a target at 20 m, and the animal's detection performance as a function of noise was determined. Using the transient form of the sonar equation, it was found that the dolphins' performances as a function of the echo signal-to-noise ratio were very similar. Dolphins process echoes like an energy detector with an integration time of approximately 264 μs. The results from the various experiments conformed to the Urkowitz [Proc. IEEE **55**, 523–531 (1967)] energy detector model. The dolphin detection sensitivity was also found to be approximately 6 to 8 dB lower than that of an ideal or matched-filter receiver.

10:51

PP10. Detection threshold calculations for Weibull interference. Joel W. Young (BBN Systems and Technologies Corporation, 4015 Hancock Street, Suite 101, San Diego, CA 92110) and Peter Cable (BBN Systems and Technologies Corporation, Union Station, New London, CT 06320)

The Weibull probability density function (pdf) has been used to represent the noise-only output of an envelope detected matched filter when the input noise contains non-Gaussian, high amplitude components. This situation may occur in active sonar when the total interference is dominated by reverberation. The familiar Rayleigh pdf associated with Gaussian input noise is a member of the Weibull family with a shape factor of 2. If the threshold for detection is based on an assumed Rayleigh density and the noise changes to Weibull with a shape factor less than 2, the false alarm rate will increase dramatically even if the average noise power is unchanged. Therefore, in order to maintain a constant false alarm rate (CFAR), the threshold must be raised. This will result in an increase in the signal to noise ratio required to achieve a specific probability of detection (P_d). It is necessary to determine the signal plus noise pdf to quantitatively evaluate this effect on detectability. An integral formulation of this function has been derived by Eckstrom, and it has been numerically integrated for a few cases by Nilsson and Glisson. The present authors have developed a much simpler approximate procedure for computing receiver operating characteristic (ROC) curves for a linear detector in Weibull noise that does not require numerical integration. This method is based on the premise that the signal plus noise pdf is not strongly affected by the Weibull shape parameter, at least over the $10\% < P_d < 90\%$ range. In this paper, the details of this approximate technique are presented, and some comparisons with more exact calculations are provided.

11:03

PP11. The use of the trans-spectral coherence technique, to separate a signal from noise. P. G. Vaidya and Mike Anderson (Mechanical and Materials Engineering Department, Washington State University, Pullman, WA 99164)

A new concept of the trans-spectral coherence has been developed. Consider a narrow-band signal A, with a center frequency $f1$, and a signal B, with a center frequency $f2$. Now, if a nonlinear filter could be found that converts the signal A into a signal C at $f2$, the trans-spectral coherence between A and B is the ordinary coherence between C and B. If the two signals A and B are derived from a common source through a nonlinear interaction, this kind of coherence is nonzero. Most speech signals, signals from musical instruments, and some signals from industrial noise sources, can be expected to possess nonzero trans-spectral coherence. In this paper, it has been conjectured that, if sound from a nonlinear source is mixed with random noise, the trans-spectral technique can be used to separate the signal from noise.

11:15

PP12. A theory for the use of windows in cepstral processing. Chris R. Fuller (Department of Mechanical Engineering, Virginia Tech, Blacksburg, VA 24061) and Jay C. Hardin (Aeroacoustics Branch, NASA Langley Research Center, Hampton, VA 23665)

Cepstral processing is often used to recover the spectrum of signals in which echoes or multipath propagation is present. Often the form of removal of echo information from the cepstral domain takes the form of a boxcar window. However, there is presently no theory to predict the influence of a cepstral window on the form of the recovered spectrum. Consequently, the use of a cepstral window is often based on a purely qualitative procedure. This paper introduces a theory that predicts the effect of a boxcar cepstral window on recovered spectra. The theory demonstrates that the actual spectrum can be recovered from the windowed spectral estimate via a purely multiplicative frequency-dependent correction factor. The theory is then applied to a signal often encountered in practice, consisting of an echoed narrow-band peak on noise. The bounds over which the cepstral window can be used with reasonable accuracy are established. Numerical simulations are then used to test the accuracy of the theory and shown to be in good agreement. [Work supported by NASA Langley.]

11:27

PP13. The application of digital filters to the de-Dopplerization of acoustic signals. Stewart A L. Glegg and Jean Marc Mouches (Department of Ocean Engineering, Florida Atlantic University, Boca Raton, FL 33431)

When acoustic measurements are made of a moving vehicle by a stationary observer, the Doppler frequency shift has two detrimental effects on the interpretation of the measured spectra. First, the spectra are smeared by the change in Doppler factor during the vehicle pass-by, and, second, the motion of the source introduces a phase shift in the signals that causes errors in source location measurements. The measured signals can be corrected back to source time if a moving time-delay correction is applied, and this overcomes both the problems mentioned above. However, when the signals are sampled digitally at equal intervals in observer time, this time-delay correction requires an estimate to be made of the signal level between samples. This can be achieved by using a digital filter with time varying coefficients, which estimates the signal from at least two adjacent samples. Results will be given that show how this technique has been applied to source location measurements on moving highway vehicles.

THURSDAY MORNING, 17 NOVEMBER 1988 KOHALA/KONA ROOM, 8:00 TO 11:45 A.M.

Session QQ. Musical Acoustics III: Musical Instruments East and West I: Pianos and Percussion Instruments

Thomas D. Rossing, Cochairman
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Hideo Suzuki, Cochairman
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Chairman's Introduction—8:00

Invited Papers

8:05

QQ1. Control of inharmonicity by the addition of a point mass. Hideo Suzuki (Ono Sokki Company, Ltd., 2-4-1 Nishi-shinjuku, Shinjuku-ku, Tokyo, 163 Japan)

A method for controlling the inharmonicity of a stiff string by adding a lumped mass (or masses) was originally suggested by F. Miller, Jr. [F. Miller, Jr., J. Acoust. Soc. Am. **21**, 318–322 (1949)]. In the present paper, the same idea is reviewed in a more precise manner. The effect of the stiffness of the string on its inharmonicity is discussed for different support conditions (simple-simple, simple-fixed, and fixed-fixed conditions). The simple-simple support condition gives the largest inharmonicity (deviation from a harmonic series), even though the differences are small in most cases. The theoretical and experimental results for the resonance frequencies of a string in the fixed-fixed condition show very good agreement with each other. For a middle C string, a mass of 0.2–0.4 g, when attached 13–9 mm from one end of the string, reduces the inharmonicity significantly.

8:35

QQ2. Measuring piano hammers and modeling their interaction. Donald E. Hall (Physics Department, California State University, Sacramento, CA 95819-2694) and Anders Askenfelt (Royal Institute of Technology, S-100 44 Stockholm, Sweden)

Comparison of measured piano string vibration spectra with predictions indicates that modeling the hammer with a simple mass and Hooke's law spring is not very satisfactory [Hall and Askenfelt, J. Acoust. Soc. Am. **83**, 1627 (1988)]. Measurements have been made on additional hammer sets to see if values of the exponent p in a nonlinear force law $F = K\xi^p$ can be connected either with hammer quality or with other aspects of piano design and history. The corresponding theory of hammer/string interaction requires solution of nonlinear equations of motion, and has previously been done only for a few isolated examples. Steps toward a flexible general computer program that can explore the dependence of string spectra upon striking parameters with nonlinear hammers will be discussed.

9:05

QQ3. Fundamental theory and computer simulation concerning the decay characteristics of piano sound. Isao Nakamura (Department of Computer Science and Information Mathematics, University of Electro-Communications, Chofu, 182 Japan)

It is shown by theory and computer simulation that the decay characteristics of piano sound are dependent upon the degree of coupling between the strings. String and soundboard vibration are calculated using an equivalent circuit. Each string is comprised of resonance circuits that correspond to partials, and the behaviors of individual partials are described independently of each other by these circuits. When the most simplified version of only one partial is considered, the equivalent circuit is expressed by the two resonant circuits coupled with the soundboard impedance. The degree of coupling between the two strings is dependent on the ratio of two constants: the degree of mistuning and the ratio of soundboard impedance to string impedance. If the former is smaller than the latter, the two strings are closely coupled with each other and a double decay characteristic results. If the relation between the two constants is reversed, the coupling is loose and a beat type of decay characteristic results. The decay characteristics are changed by the above two constants. The results show that the degree of tuning and the characteristics of the strings and the soundboard determine the decay characteristics of piano sound.

9:35

QQ4. Acoustics of Eastern and Western bells, old and new. Thomas D. Rossing (Department of Physics, Northern Illinois University, DeKalb, IL 60115)

Bells have belonged to nearly every culture in history. The oldest surviving bells from the Near East and China are thought to be over 3000 years old. Many ancient Chinese bells, by virtue of their oval shape, have two distinctly different strike notes, separated in pitch by about a major or minor third. Bells developed as Western musical instruments in the 15th and 16th centuries when bell founders learned how to tune their partials harmonically. The acoustical behavior of Western church bells, carillon bells, and handbells are compared to those of ancient Oriental bells. Recently carillon bells with a major third replacing the traditional minor third partial have been developed [A. Lehr, Music Percept. **4**, 267 (1987)].

10:05–10:15
Break

THURSDAY AM

10:15

QQ5. Acoustical properties of ancient Chinese musical qing instrument. Chen Tong (Institute of Acoustics, Academia Sinica, P.O. Box 2712, Beijing, People's Republic of China)

The qings were important percussion musical instruments in ancient China. A qing is a stone plate with a special form. The vibration of a qing is studied both theoretically and experimentally in this paper. The vibrational modes of a qing have been calculated by using a finite element method in accordance with the experimental results. As compared with a rectangular plate, the frequency relations between the overtones are varied with the changes of shape of the qing. From the analysis, it is possible to speculate upon the idea of design of the qing in ancient time. In order to confirm the idea, the sound of a qing has been simulated by computer and judged subjectively. An empirical formula for the calculation of the pitch frequency of a qing sound is given. Some examples show the musical scale of ancient qing groups.

10:45

QQ6. Pitch of a set of ancient Chinese bells: Piao-shi Bian Zhong. Junji Takahashi (Laboratory of Musical Acoustics, Music Research Institute, Osaka College of Music, Toyonaka, Osaka, 561 Japan)

The precise pitch and size are measured for a set of 12 ancient Chinese bells, named "Piao-shi Bian Zhong" and made probably in 404 B.C., which have been preserved at the Sen-Oku Hakko Kan Museum in Kyoto. The complete set consists of 14 bells, including the 12 bells measured in this paper and two other bells preserved at the Royal Ontario Museum in Canada. Zhong, a kind of Chinese bell with an almond-like cross section, emits two tones according to the position struck. The bells are all constructed in a similar shape except for the thickness. The frequencies of the 12 lower tones were found to be inversely proportional to the squared distance between the two tails of the lateral spines, called "xian," of the bells. This was the same with the upper tones. This relation suggests that ancient craftsmen should probably have taken advantage of such an empirical formula to determine the size of each bell to produce two specified pitches of a musical scale.

11:15

QQ7. Harpsichord modal behavior. William R. Savage[a] (Department of Physics and Astronomy, University of Iowa, Iowa City, IA 52242), Kenneth Marshall (Uniroyal Goodrich Tire Company, Brecksville, OH 44147), Edward L. Kottick (School of Music, University of Iowa, Iowa City, IA 52242), and Thomas Hendrickson (Department of Physics, Gettysburg College, Gettysburg, PA 17325)

The modal behavior of the harpsichord in the acoustics laboratory at the University of Iowa was determined by percussing approximately 600 points on the external surfaces of the instrument with an instrumented hammer. Both the input from the hammer and the response of the harpsichord were recorded. The data were digitized and the response of each tapped point was converted to a numerical value. These data have been processed into a computer-generated, three-dimensional video tape showing the harpsichord's vibrating modes from 0–500 Hz. Of particular interest is the way in which all the external structural elements of the harpsichord can be shown to vibrate at some point. The results of this work agree closely with the published [E. L. Kottick, Galpin Soc. J. **48**, 55–77 (1985)] and unpublished data developed by the authors of this paper.
[a] William Savage died 28 May 1988.

11:30

QQ8. Optical modal analysis compared to modal analysis with digital electronic equipment. Erik V. Jansson (Department of Speech Communication and Music Acoustics, Royal Institute of Technology, S-100 44 Stockholm, Sweden) and Nils-Erik Molin (Luleå University of Technology, S-951 87 Luleå, Sweden)

Hologram interferometry and speckle interferometry have introduced powerful tools for "optical modal analysis." The vibration amplitudes at a single frequency are recorded simultaneously for all points of a surface. The excitation position and frequency are adjusted to give a normal mode by watching the vibration patterns in real time. In the common modal analysis with digital electronic equipment, the vibrations of a single point as a function of time are recorded for all frequencies simultaneously via the impulse response. Impulse responses for a large number of points are measured, analyzed, and recalculated to extract the normal modes. Investigations of two modes of a complicated object (a violin) using both methods showed differences. The position of the dominating antinodes agreed, but the distribution of the vibration amplitudes and especially the nodal lines did not always agree. The differences seem to depend mainly on methods. The differences imply that great cautiousness is motivated in interpreting unknown vibration modes of objects with complicated structures and boundaries. The real time observation with the optical modal analysis offers hereby advantages.

THURSDAY MORNING, 17 NOVEMBER 1988 — WAIANAE ROOM, 8:00 TO 11:52 A.M.

Session RR. Physical Acoustics V and Bioresponse to Vibration III: Biomedical Ultrasound

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Tokyo, 158 Japan

Invited Papers

8:00

RR1. Field measurements of ultrasonic diagnostic equipment. Masao Ide (Department of Electronics and Communication Engineering, Musashi Institute of Technology, Setagaya-ku, Tokyo, 158 Japan)

Recent progress and popularization of ultrasonic diagnostic equipment are remarkable, and the standardization of this equipment, especially of its acoustic output which is related to bioeffects, has become important. IEC began by dealing with the standardization of the measuring methods of acoustic fields. Studies on methods of measuring acoustic fields of diagnostic equipment have been made by such researchers as R. C. Preston, G. R. Harris, P. A. Lewin, and M. Ide, and others [IEEE Trans. Ultrason. Ferroelec. Freq. Contr. **35** (1988)]. Since the field irradiated from the diagnostic equipment is pulsed ultrasound, many parameters have to be considered. The most useful measuring method, at present, is the planar scanning method using a small hydrophone. Using this method, the values of the main parameters can be obtained relatively easily. This paper mainly shows this type of measuring method in the laboratory. By scanning a small hydrophone in water in three dimensions, the acoustic pressure at each point is measured and stored in a computer, and then all the spatial and temporal parameters are computed. Examples of measured acoustic data from ultrasonic diagnostic equipment merchandised in Japan are shown.

8:20

RR2. Absorption and attenuation in biological media. Frederick W. Kremkau (Ultrasound Center, Bowman Gray School of Medicine, Wake Forest University, Winston-Salem, NC 27103)

Attenuation of ultrasound in tissues is due primarily to absorption at the macromolecular level and is dominated by proteins. Protein absorptions are large compared to those for the amino acids of which they are made. Conversion of structured native proteins to random chains reduces absorption significantly in linear proteins but causes only small changes in globular proteins. Phosphate buffer ions dramatically increase absorption in sugars and amino acids but make only small changes in polysaccharides and proteins. Protein absorption increases nonlinearly with concentration. Organized cellular components (plasma membrane and nucleus) have higher absorption than the cell as a whole. Current working hypotheses regarding biomolecular absorption include: (1) The underlying absorption as a function of molecular weight is similar to that for dextran, (2) secondary structure contributes to absorption, (3) solvation contributes to absorption through proton transfer and perturbation of bound-free water equilibria, (4) tertiary structure inhibits solvation absorption in globular proteins, and (5) macromolecular interactions in high concentration, aggregation, and cross-linking conditions contribute to absorption. [Work supported by NCI and NIGMS, DHHS.]

8:40

RR3. Velocity of sound in biological materials. Edwin L. Carstensen (Departments of Electrical Engineering, Biophysics and the Rochester Center for Biomedical Ultrasound, University of Rochester, Rochester, NY 14627)

The phase velocity of longitudinal waves in biological fluids and tissues with high water content exceed the velocity of sound in water roughly in proportion to the solid content of the material. In general, fatty tissues have lower and bone higher sound speeds than water. With the exception of water, the sound speeds of most of the components of tissues have negative temperature coefficients. Dispersion in the velocity of sound is related to the distribution of relaxing elements in the material and, hence, to the absorption. Since the velocity is a function of the pressure, sound propagation is a nonlinear process. Biological materials, in general, are more nonlinear than water. High-amplitude sound waves become distorted as they propagate and may form shocks that have been associated with biological effects such as excess absorption and fragmentation of kidney and gall stones.

THURSDAY AM

9:00

RR4. Analysis of echoes from a sphere that includes the directivity of a transmitter and a receiver. Mitsuhiro Ueda and Eishi Morimatsu (Research Laboratory of Precision Machinery and Electronics, Tokyo Institute of Technology, Nagatsuta, Midori-ku, Yokohama, 227 Japan)

Echoes from a sphere are important in biomedical ultrasound in many respects. For example, a sphere is used as a test object for evaluating the resolving power of pulse echo systems, and it shows a characteristic echo pattern such as the cometlike pattern on *B*-mode images. The speckle pattern on *B*-mode images can also be modeled by suspension of the spherical particles. The analysis of echoes from a sphere, however, has been limited to the case of plane-wave illumination. Consequently, it has been rather difficult to apply the results of this analysis to medical ultrasound since focused beams have been used in this area. In this report, the plane-wave theory for echoes from a sphere is extended to the case where arbitrary transmitting and receiving ultrasonic transducers are used, and the effect of the directivity of transducers is analyzed both on the waveform of echoes and on the backscattering coefficient measurement of suspension.

9:20

RR5. An ultrasonic scattering characterization of tissue. Robert C. Waag (Departments of Electrical Engineering and Radiology, University of Rochester, Rochester, NY 14627)

Average differential scattering cross section is obtained from measurements by a normalization that accounts for temporal waveforms and spatial apertures. A model is employed to express the average differential scattering cross section as a linear combination of the power spectrum of the medium compressibility variations, the power spectrum of density variations, and the cross power spectrum of compressibility and density variations. Since the scale factors depend on scattering angle and temporal frequency, known values of the average differential scattering section at different scattering angles and temporal frequencies corresponding to the same spatial frequency, which is considered to be the argument of the average differential scattering cross section and the power spectra, are used to calculate each of the three power spectra. Results are shown for model random media and calf liver. The results indicate that determinations of average differential scattering cross sections and the power spectra of scattering medium variations can be made under practical conditions and also imply that density variations contribute significantly to scattering by calf liver. [Work supported by NSF, NIH, NATO, and Industrial Associates.]

9:40

RR6. Application of the acoustic nonlinearity parameter to tissue composition prediction. Robert E. Apfel and E. Carr Everbach (Department of Mechanical Engineering, Yale University, P. O. Box 2159, New Haven, CT 06520)

The objective of this research is the prediction of the composition of tissues (percent water, protein, and fat) from the results of ultrasonic measurements (sound speed and nonlinear parameter) and reasonable mixture laws (for the compressibility, density, and nonlinear parameter). These laws are a redundant set allowing for the use of an optimization routine to predict the volume fractions of a three-component mixture. When applied to the data of others for fluid mixtures and tissues, reasonable agreement with known compositions has been achieved. This methodology has now been applied to tissue phantoms composed of known amounts of gelatin, water, and fat. One crucial requirement for the theory's application is a convenient apparatus that gives accurate, repeatable data for the nonlinear parameter, as has not always been possible with finite amplitude and traditional thermodynamic methods. In these experimental procedures, a commercial instrument is adapted employing a phase-locked loop to a modified thermodynamic method first used by Emery *et al.* [J. Phys. (Paris) **11**, 231–234 (1979)]. This apparatus permits rapid and precise measurements. Results (and limitations) of recent measurements and comparisons with theoretical predictions will be presented. [Work supported by the U. S. National Institutes of Health through Grant R01GM30419.]

10:00

RR7. Real-time two-dimensional blood flow imaging using the ultrasonic Doppler effect. Chihiro Kasai (Aloka Company, Ltd., 6-22-1 Mure, Mitaka, 181 Japan)

A newly developed two-dimensional blood flow imaging system that uses the ultrasound Doppler effect is presented, in which blood flow behavior in a given cross section of a living organ is displayed in real time. To perform this, an autocorrelation technique is employed. In the technique, ultrasonic pulses are transmitted several times in the same direction and the phase shifts of the returning echoes from moving blood cells are extracted. The blood flow data obtained are displayed in color on a TV screen superimposed on tissue images that are displayed in black/white. The flow direction, speed, and variance are expressed by differences in color and brightness. Experiments were conducted with a mechanical and an electrical scanner, and good agreement with the theory was obtained. Studies on clinical significance have also been carried out at hospitals, and the usefulness of the system has been demonstrated. The system is now becoming an indispensable tool in diagnosing circulatory diseases such as heart diseases and vessel diseases.

10:20

RR8. Flow vector mapping through stream function from two-dimensional Doppler imaging. Shigeo Ohtsuki, Motonao Tanaka, and Motoyoshi Okujima (Tokyo Institute of Technology, Yokahoma, 227 Japan)

The distribution of Doppler velocity in the heart or large vessels can be observed in real time by a sector-scan or linear-scan Doppler ultrasound system. This Doppler velocity means one velocity component detected by the Doppler effect of reflected ultrasound from a moving target. Coding the magnitude of the Doppler velocity to a color, the distribution is displayed in Doppler color imaging. On the other hand, the distribution of a flow vector component on an observing plane gives more information than the Doppler color imaging. In order to derive flow vectors on a plane from Doppler color imaging, a technique to calculate stream functions on a plane was developed. The contour lines of this function are stream lines. The calculation of a stream function consists of the integration of weighted Doppler information. The distribution of a two-dimensional flow vector component can be calculated from this stream function. This technique was applied to the blood flow in the heart.

10:40

RR9. Measurement of the nonlinearity parameter *B/A* of tissue by the time difference method. Nobuyuki Endoh, Tetsuya Asahina, Masayuki Igawa, and Masahide Hamamoto (Department of Electrical Engineering, Kanagawa University, Yokohama, 221 Japan)

A method of measuring the nonlinearity parameter B/A of soft tissue *in vitro* was developed for thermodynamic phenomenon. Using a gated ultrasonic rf pulse, the propagation time through the tissue was measured at two different thickness values. The B/A of the tissue could be estimated by the thickness difference, and the propagation time difference between atmosphere and a pressurized (3 atms) condition. A fundamental experiment was carried out to confirm the validity of the method. The phase (time) difference was measured by the interference method with a two-channel synthesizer. The B/A of water was proportional to its temperature from 10 to 40 °C and was 5.1 at 20 °C. A B/A of 9.2 for a chicken liver *in vitro* was also obtained.

10:52

RR10. Ultrasonic spectroscopy in bovine serum albumin solutions. Pak-Kon Choi, Jung-Rim Bae, and Kenshiro Takagi (Institute of Industrial Science, University of Tokyo, Minato-ku, Tokyo, 106 Japan)

Ultrasonic absorption was measured in bovine serum albumin aqueous solutions (50 g/l) in the frequency range from 100 kHz to 1.6 GHz at *p*H of neutral, acid, and alkaline regions. Three experimental techniques were used to cover the wide frequency range: plano-concave resonator, conventional Bragg reflection, and high-resolution Bragg reflection (HRB) methods. At *p*H 7, the absorption spectrum fitted the relaxation curve well using the distribution function of a mirror image of Davidson–Cole. This suggests that the dominant relaxation mechanism is due to hydration. In the acid region, the relaxation peak observed at 2 MHz was attributed to the proton transfer of the carboxyl group on the glutamic and aspartic acid residues. Another relaxation peak at 200 kHz was associated with the conformational change in the albumin molecules. The two relaxation peaks in the alkaline region were ascribed to the proton transfers on the tylosyl and lysyl residues. A relaxation due to the conformational change was also indicated.

11:04

RR11. Numerical models of perfused hyperthermia phantoms. Eric J. Shrader, Phillip R. Jeuck, and Peter D. Edmonds (Mechanical and Bioengineering Research Laboratories, SRI International, Menlo Park, CA 94025)

Finite difference numerical models based on bioheat transfer [H. H. Pennes, J. Appl. Physiol. 1, 93–122 (1948)] and incomplete countercurrent heat exchange [S. Weinbaum and L. M. Jiji, J. Biomech. Eng. 107, 121–139 (1985)] have been developed and tested. Predicted performance of hyperthermia phantoms heated by ultrasound applicators will be compared with actual performance to validate the models. The models will then be used as aids in designing ultrasound applicators to produce heat deposition patterns believed advantageous for hyperthermia treatment of experimental animals and patients. The 3D models admit adiabatic or isothermal boundaries and arrays of paired, straight tubules for counter current heat exchange. Results of parametric variations will be shown. Preliminary results obtained by using as inputs the design parameters for a proposed kidney phantom perfused at 110 ml/100 g (of tissue-equivalent gel) per minute have yielded satisfactory time constants on the order of 150 s. When the simulated ultrasound applicator emitted 56 acoustic watts at 1 MHz, applied as an 8-cm-diam, unfocused Gaussian beam, the predicted steady-state, spatial maximum temperature rise was 6 °C. [Work supported by PHS Grant No. R01 CA33749 awarded by the National Cancer Institute, DHHS.]

11:16

RR12. Ultrasonic imaging of the internal vibration of soft tissue under forced vibration. Yoshiki Yamakoshi, Jun'ichi Sato, and Takuso Sato (The Graduate School at Nagatsuta, Tokyo Institute of Technology, Midori-ku, Yokohama, 227 Japan)

To estimate the dynamic properties of soft tissues, an imaging system that can display both the amplitude and phase distributions of vibration in soft tissues under forced vibration was developed. In this method, low-frequency sinusoidal vibration with a frequency from 10 Hz to 1 kHz is applied from the surface of the sample, and the movement in it is measured using simultaneously transmitted probing ultrasonic waves. Then scatterers in the sample move sinusoidally under the forced vibration, and the reflected ultrasonic waves are subject to the frequency modulation by the Doppler effect. The amplitude of vibration is estimated from the frequency modulation index of the reflected waves. This is derived from the amplitudes of the spectrum components of the output of the quadrature detector of the received signal. On the other hand, the phase of vibration is estimated from the angle of the fundamental spectrum component. Basic experiments have been carried out using a 3.0-MHz ultrasonic wave system. The accuracy of the method is evaluated and two-dimensional color maps of the amplitude and phase are shown for several phantoms, as well as tissues, *in vivo*.

11:28

RR13. Three-dimensional reconstruction of real-time ultrasonic scans. Kenneth L. Watkin (Biomedical Engineering, McGill University, Montreal, Quebec H3A 2B4, Canada)

Real-time ultrasound scan images are gathered at unequally spaced, noncoplanar intervals. Three-dimensional reconstruction of ultrasonic scan images requires information on the location and angle of the ultrasound transducer simultaneously with the ultrasound image. Using a pulsed inductive coil motion monitoring system and a real-time ultrasound scanner, the purpose of the present investigation was to compare measurements of a fetuslike phantom with the three-dimensional reconstructed image of this phantom. The output of the motion sensor and the ultrasound monitoring system were recorded simultaneously, digitized and then used to determine transducer position/angle and the edges of the fetuslike phantom immersed in water. Geometric modeling techniques including finite element mesh generation were used to reconstruct the phantom. Area and volume comparisons of the phantom and reconstructed image were conducted. Although cross-sectional area comparisons revealed similar measurements, accuracy of volume estimations were dependent on the number of slices used for the calculation. Results will be discussed in light of current geometric modeling methods and the reconstruction of ultrasonic scans.

11:40

RR14. New technique for spot-poling piezopolymer membrane hydrophones. Aime' S. DeReggi (Polymers Division, National Bureau of Standards, Gaithersburg, MD 20899) and Gerard R. Harris (Center for Devices and Radiological Health, Food and Drug Administration, Rockville, MD 20857)

One characteristic that poses a lower limit to the size of existing spot-poled membrane hydrophones is that the active region is always larger than the electroded region. The difference is due in part to the fringe poling fields that surround the electrodes and proximal portions of the leads. This effect makes it difficult to construct hydrophones having dimensions less than $\simeq 0.5$ mm. A potential way around this problem is to reverse the standard poling and plating process. That is, the film is first poled and then the metal electrodes and leads are added. This approach has been applied using small (<0.1-mm tip diameter) needles as poling electrodes. Piezopolymer film was mounted on support rings and a central spot was poled at room temperature. Electrodes and leads were vacuum deposited, and sensitivity and directional response measurements were made. In this presentation, details of the poling process will be described and measurement results will be discussed.

Session SS. Psychological Acoustics IV: Pitch, Timbre, the Coding of Complex Sounds, and Masking (Poster Session)

Otoichi Kitamura, Cochairman
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Neal Viemeister, Cochairman
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Contributed Papers

All posters will be displayed from 8:00 a.m. to 12:00 noon. To allow contributors an opportunity to see other posters, contributors of papers SS1 through SS9 will be at their posters from 8:00 to 9:15 a.m., contributors of papers SS10 through SS18 will be at their posters from 9:15 to 10:30 a.m., and contributors of papers SS19 through SS27 will be at their posters from 10:30 to 11:45 a.m.

SS1. Detection of increments and decrements in modulation depth of SAM noise. D. Wesley Grantham (Bill Wilkerson Hearing and Speech Center, 1114 19th Avenue South, Nashville, TN 37212) and Sid P. Bacon[a] (Division of Hearing and Speech Sciences, Vanderbilt University School of Medicine, Nashville, TN 37232)

Difference limens (DLs) for the modulation depth of a sinusoidally amplitude-modulated (SAM) wideband noise were measured using a two-interval forced-choice adaptive procedure. For increment detection the reference modulation depth (m_r) was 0.0, 0.25, 0.50, or 0.75. For decrement detection, m_r was 0.45, 0.60, 0.75, or 0.90. Modulation frequency was 2, 4, 64, or 512 Hz. The DL is defined as $10 \log|m_{thr}^2 - m_r^2|$, where m_{thr} is the modulation depth just discriminable from m_r. For $m_r = 0.0$, the DLs (or "absolute" modulation thresholds) varied from -25 to -10 dB as modulation frequency varied from 2–512 Hz. As m_r increased, the increment DL increased monotonically for each frequency from its absolute detection level to about -6 dB at $m_r = 0.75$. These data are in good agreement with those of Wakefield and Viemeister [J. Acoust. Soc. Am. Suppl. 1 **72**, S90 (1982)]. The decrement DLs overlay and extended the increment DL functions, although the DLs for the highest m_r (0.90) decreased slightly (-10 dB). In a separate manipulation, no effect on the modulation onset phase on modulation depth discrimination was found. [Work supported by NIH.] [a] Current address: Department of Speech and Hearing Science, Arizona State University, Tempe, AZ 85287.

SS2. Interference of modulation rate discrimination of pure tones. William A. Yost, Stanley Sheft, and Jane Opie (Parmly Hearing Institute, Loyola University, 6525 North Sheridan Road, Chicago, IL 60626)

Modulation rate discrimination (Δfm) thresholds were obtained for sinusoidally amplitude modulated (SAM) pure tones. The carrier frequencies were 1000 and 4000 Hz and the modulation rates were 5, 10, 20, and 50 Hz. The Δfm thresholds of one carrier (the probe) were also obtained when the other carrier (the masker) was presented simultaneously. The masker was either SAM or was unmodulated. In all conditions the depth of SAM was 100%. This experiment was designed to determine how much the modulated masker interferes with the discrimination of modulation rate and how this interference changes as a function of the difference in modulation rate between the masker and probe. It has been shown previously [Yost and Sheft, J. Acoust. Soc. Am. Suppl. 1 **83**, S35 (1988)] that a modulated masker interfered significantly with a listener's ability to detect SAM of the probe when the masker and probe were modulated at the same rate. This modulated detection interference (MDI) was reduced when the masker was modulated at a different rate than the probe. These earlier results suggested that there might be auditory channels "tuned" to different rates of amplitude modulation. The present experiment was designed to further test the concept of AM tuned channels. [This research was supported by the NINCDS.]

SS3. Dynamic perception of tones simultaneously modulated in both frequency and amplitude. Takashi Tsumura (Department of Acoustic Design, Kyushu Institute of Design, 4-9-1 Shiobaru, Minami-ku, Fukuoka, 815 Japan)

Auditory processing in the dynamic perception of tones with simultaneous frequency modulation (FM) and amplitude modulation (AM) was investigated. Thresholds for the detection of target modulation (FM or AM) were measured with the respective nontarget modulation (AM or FM) kept well above threshold. The carrier frequencies of tones below 1 kHz were used. For FM and AM having the same sinusoidal modulation frequencies below 20 Hz, the thresholds of the target modulation changed as a function of the phase difference between the target and nontarget modulations. When the frequencies of the FM and AM were different from each other, the thresholds of the target modulation were nearly the same as those obtained for a tone with the target modulation alone. When the frequency and amplitude continuously increased or decreased in a short tonal duration, there was little difference in the threshold of the target modulation between the cases for increase and decrease in nontarget modulation. These results suggest the existence of channels selectively tuned to the periodicity of FM and AM in the auditory pathways. [Work supported by a Grant in Aid for Scientific Research (No. 62510059) from the Ministry of Education.]

SS4. Effects of manipulating the spectral shape of reproducible noise samples on detection judgments. R. H. Gilkey and B. D. Simpson (Central Institute for the Deaf, 818 South Euclid, Saint Louis, MO 63110) and A. M. Hammoud (Central Institute for the Deaf and Washington University, Saint Louis, MO 63130)

The single-interval/yes–no detection judgments of subjects to a subset of ten noise-alone and ten signal-plus-noise samples from the 300 waveforms examined by Gilkey and Meyer [J. Acoust. Soc. Am. Suppl. 1 **82**, S92 (1987)] were investigated further by manipulating the spectral shape of the samples. The spectral shape was "modulated" by either increment-ing or decrementing the level in a 47-Hz band centered around one of seven center frequencies ranging from an octave below to an octave above the signal frequency. The waveform on each trial within a block was randomly sampled without replacement from a large set of modulated and unmodulated waveforms. Thus the experience of the subject was assumed to be comparable to that with truly random noise. Detection judgments are affected by changes in the shape of the spectrum that are well outside the critical band centered at the signal frequency. The results are discussed in relation to models of auditory masking. [Work supported by NSF and AFOSR.]

SS5. Critical band distortion analysis. Edward Cudahy (Lexington Center, 30th Avenue and 75th Street, Jackson Heights, NY 11370), Harry Levitt (Center for Research in Speech and Hearing Sciences, City University of New York, 33 West 42nd Street, New York, NY 10036), and H. Cynthia Link (Lexington Center, 30th Avenue and 75th Street, Jackson Heights, NY 11370)

Current measurement techniques typically focus on distortion produced by clipping or intermodulation of pure tones. There is little information on the relation between these measurements and the perception of distortion. The present study had two purposes. First, to develop a general method for measuring distortion. Second, to determine the association between this physical measurement and the perception of distortion for normal hearing listeners. Results suggest that a critical band analysis of distortion can be useful for predicting thresholds for distortion, at least for simple stimuli. [Research supported by NIDRR.]

SS6. A minimum discriminable bandwidth test for critical bandwidth estimation. Robert D. Celmer (College of Engineering, University of Hartford, West Hartford, CT 06117) and Gordon R. Bienvenue (Communications Department, State University of New York, New Paltz, NY 12561)

A test procedure has been developed through the use of a modified method of limits technique for the direct measurement of critical bandwidth. A tonal complex with an initially subcritical bandwidth is presented to a listener. The test signal then discretely widens in bandwidth with time; each bandwidth has a duration of approximately 800 ms. Subjects indicate at which bandwidth they perceive a change in the sound. The minimum bandwidth at which a listener can discriminate a difference in the tonal complex is taken to be that subject's critical bandwidth (cf. Scharf, 1970). The tonal complexes were generated using a digital signal processing algorithm, with the value of the initial bandwidth randomized to reduce learning effects. Results of the test performed on normal and sensorineural hearing impaired listeners are presented. Implications for evaluating sensorineural hearing impairment are discussed.

SS7. Frequency effects for multicomponent maskers with high spectral uncertainty. Donna L. Neff and Brian P. Callaghan (Boys Town National Institute, 555 North 30th Street, Omaha, NE 68131)

Simultaneous maskers composed of a few sinusoidal components can produce considerable masking if the component frequencies are unpredictable. This experiment examined how components in particular frequency regions influenced performance. Thresholds were measured for a 1000-Hz signal presented simultaneously with a masker composed of 2, 4, 6, 10, 50, or 100 components. Components were drawn from a 300–3000-Hz range, excluding the critical band around 1000 Hz. When sufficient masking was produced by uncertainty, limiting components to the high- or low-frequency side of the signal, or widening the notch around the signal, had little effect. Decreasing uncertainty reduced thresholds for maskers limited to high-frequency components, but not for those limited to low frequencies. Paradoxically, forcing components into narrower bandwidths around the signal could reduce masking by about 5 dB. Large individual differences on this task were not well predicted by measures of auditory filter shape with notched-noise maskers. [Work supported by AFOSR.]

SS8. Detection of a silent gap in single components of a multitonal complex. David M. Green and T. G. Forrest (Psychoacoustics Laboratory, Department of Psychology, University of Florida, Gainesville, FL 32611)

Complex multitonal waveforms were generated using 21 equal-amplitude sinusoids. The tones range in frequency from 200–5000 Hz, and the frequency ratio of the successive components was constant (equal-logarithmic intervals). The listener's task was to detect the silent gap in a single component of the complex. The gap occurred either in the temporal center of a 500-ms burst or at a randomly chosen temporal location. For both conditions, gap threshold decreases with component frequency from about 20–50 ms at the lower frequencies to about 2–5 ms at the highest frequency. It is somewhat easier to hear a gap if it is present in more than a single component. Synchronized gaps are slightly easier to hear than asynchronous gaps, even if the asynchronous gaps occur at predictable times. [Research supported by AFOSR and NSF.]

SS9. Discrimination of recycled word-length sequences. James A. Bashford, Jr. and Richard M. Warrren (Department of Psychology, University of Wisconsin—Milwaukee, Milwaukee, WI 53201)

Earlier studies in this laboratory have dealt with the ability to distinguish between permuted orders using recycling or locked sequences of three of four items. More complex "Watson" sequences, presented as single bursts, typically of ten items, have been used to examine discrimination involving "word-length" patterns. The present study combines elements of both techniques, investigating the ability to discriminate minimal changes of order within recycled word-length sequences of 40 ms items: sinusoids (experiment 1); vowels (experiment 2); and frozen noise segments (experiment 3). Listeners made ABX judgments for pairs of ten-item sequences differing only in the ordering of two contiguous items. Pattern pools of 48 sequence pairs were employed in each experiment. This hybrid technique of recycling word-length sequences permits ready discrimination of permuted order: For example, naive listeners presented with tone sequences performed at levels well above chance at the outset of training and approached 96% accuracy in less than 100 trials. Special aspects of results obtained with sequences of vowels and frozen noise will be discussed. [Work supported by AFOSR.]

SS10. Detection of changes in frequency- and time-transposed auditory patterns. Gary R. Kidd and Charles S. Watson (Department of Speech and Hearing Sciences, Indiana University, Bloomington, IN 47405)

It is well known that listeners can recognize familiar melodies that have been transposed in pitch or played in a different tempo [e.g., J. C. Bartlett and W. J. Dowling, J. Exp. Psychol.: Hum. Percept. Perform. **6**, 501–515 (1980); R. Francès, *La perception de la musique* (J. Vrin, Paris, 1958)]. However, the effect of transposition on the recognition of unfamiliar auditory patterns that do not have the pitch-material or stuctural constraints of musical patterns has not been investigated. Listeners' abilities to detect frequency changes in transpositions of randomly generated five-tone patterns were examined using a same–different adaptive-tracking procedure. On each trial a standard pattern was followed by a comparison pattern that was transposed in frequency, time, or both dimensions. The magnitude of a frequency difference introduced in a single tonal component was adjusted from trial to trial, based on the listener's performance. Frequency and time transpositions consisted of increases in frequency and duration, respectively, ranging from 12%–100%. Listeners were told to ignore the changes in absolute frequency and time and make their judgments on the basis of the pattern of intervals. Increases in both frequency and temporal transposition lead to decrements in performance. However, on the average, frequency transposition yields more severely degraded performance than temporal transposition in the range of conditions studied. [Work supported by AFOSR and NIH.]

SS11. Effect of attention on hearing ability. Shin'ichi Suzuki, Tsuyoshi Usagawa, and Masanao Ebata (Faculty of Engineering, Kumamoto University, 2-39-1 Kurokami, Kumamoto, 860 Japan)

The temporal characteristics of attention in auditory signal processing are studied experimentally. The effect of information about the timing of a tone's occurrence on its threshold and DL (difference limen) was examined. When uninformed about the timing of tone occurrence, subjects must keep their attention throughout the experimental period. This concentration makes the subject so tired that he often fails to detect the tone. In the present study, the test tone was presented at fixed and random timings. The threshold was measured using a 1-kHz pure tone at seven levels, and the DL was measured using an overmodulated AM tone by the method of constant stimuli. The results show that the information on the timing of the tones lowers thresholds and reduces DLs. This phenomenon can be regarded as a result of an improvement in the S/N ratio because of the correlation between the input auditory signal and the knowledge of its timing.

SS12. Attentional filters and frequency uncertainty in the detection of one or more tones. Robert S. Schlauch and Ervin R. Hafter (Department of Psychology, University of California, Berkeley, CA 94720)

A probe-signal method [G. Z. Greenberg and W. D. Larkin, J. Acoust. Soc. Am. **44**, 1513–1523 (1968)] is used to study the listening bands for tonal signals in noise. In a two-interval, forced choice paradigm, *cues* mark the frequency of a randomly chosen *target* signal on about 70% of the trials while *probe* signals on the remaining trials have frequencies that differ from the cues by ratios ranging from 0.7–1.4. Based on pretesting, levels are set to produce 90% correct detections when the signal's frequency matches the cue; this is true whether or not the target is roved from trial to trial. Listening bands are studied by plotting all of the data relative to the target frequencies. In a test of a theory that postulates that bandwidth may vary in response to uncertainty [Johnson and Hafter, Percept. Psychophys. **28**, 143–149 (1980)], two or more equally likely targets are cued on every trial, allowing for simultaneous measurement of multiple listening bands.

SS13. Development of a model for multidimensional identification experiments. Louis D. Braida (Research Laboratory of Electronics, Masschusetts Institute of Technology, Cambridge, MA 02139)

When several aspects of a stimulus are identified simultaneously, the pattern of errors made by an observer is generally not well described by a unidimensional Thurstonian decision mode. Although such data are often interpreted with the aid of multidimensional scaling techniques, this approach describes only very crude aspects of confusion matrices. A multidimensional generalization of the Thurstonian decision model has been developed that permits relatively precise predictions of error patterns to be made from parameters that describe sensitivity and bias. This model has been applied to identification experiments using vibrotactile stimuli varying over large ranges of intensity, frequency, and contactor area [Rabinowitz *et al.*, J. Acoust. Soc. Am. **82**, 1243–1252 (1987)]. Analysis reveals, for example, that sensitivity for a given stimulus dimension is generally reduced when both that dimension and another dimension are identified. This reduction is partially responsible for the failure of multidimensional stimuli to transmit as much information as performance for the separate dimensions would suggest. Also, for vibrotactile stimuli, the perceptual coordinates corresponding to frequency and contractor area are relatively orthogonal while that corresponding to intensity is skewed relative to these. [Work supported by NIH.]

SS14. Stream segregation—source grouping versus peripheral channeling. William Morris Hartmann and Douglas Johnson (Department of Physics, Michigan State University, East Lansing, MI 48824)

The prevailing view is that auditory stream segregation should be understood as fundamentally teleological. Two tones are segregated into the same stream if, according to Gestalt grouping rules, they probably derive from the same acoustical source. A problem with this point of view is that there are many different stimulus features that might serve equally well to identify and group sources, but not all of these features are equally effective in promoting stream segregation. Using an interleaved melody identification paradigm, the ability of listeners to segregate streams based upon ten different kinds of stimulus features was measured. The results show that tones that excite different peripheral channels of the auditory system, based upon differences in octave range, ear presentation, or spectral envelope, can be separately streamed with very high efficiency. Tones that excite primarily the same peripheral channels, but that may be identified as different sources based upon level, duration, amplitude envelope, or interaural phase difference, can be separately streamed, but with markedly less efficiency. Tone sequences that employ irregular rhythm or added noise cue bands do not promote stream segregation. [Work supported by the NIH.]

SS15. Pitch of complex tones. Masanao Ebata, Youichi Matsuyama, Shin'ichi Suzuki, and Tsuyoshi Usagawa (Faculty of Engineering, Kumamoto University, 2-39-1 Kurokami, Kumamoto, 860 Japan)

There has been much research on low pitch, but the mechanism of the perception of low pitch has not yet been made clear. In this paper, the characteristics of the pitch of complex tones were compared with those of pure tones. The effect of duration on pitch discrimination for complex tones was almost the same as that for pure tone, but the slope of the DL curve as a function of the duration was a little slower for the complex tone than for the pure tone in the region of shorter duration. The pitch shift due to the auditory fatigue observed for the pure tones was also observed for the complex tones. The effect of a preceding tone on the pitch of a complex tone was also observed. The pitch of the complex tone was influenced by a preceding tone with the same frequency as one of the components of the complex tone. Based on these observations, the process of pitch extraction for complex tones was discussed.

SS16. Informational processing of complex sound. Robert A. Lutfi (Waisman Center, University of Wisconsin, Madison, WI 53706)

Normal-hearing listeners discriminated among complex, nonspeech sounds in which the *difference* to be discriminated was randomly varied from trial to trial. All other physical parameters of the sounds were fixed within a block of trials. On each trial, the listener heard two sounds (e.g., tone complexes). The values of the variable parameter (e.g., tone frequencies) were drawn from two normal distributions differing only in mean. The listener's task was to identify the sound having the higher mean value of the variable parameter. Discrimination performance was found to be largely independent of the particular physical dimensions along which the sounds varied. Rather, performance appeared to depend primarily on information content of the sounds. Information content was defined in terms of a stimulus equivocation factor that was derived from the data. Based on this model, transmitted information was estimated to be between 1.0–3.0 bits. [Work supported by AFOSR.]

SS17. Comodulation masking release with listening-condition uncertainty. Beverly A. Wright and Dennis McFadden (Department of Psychology, University of Texas, Austin, TX 78712)

Detectability of a 1250-Hz tonal signal, masked by 50-Hz-wide noise bands centered at 850, 1050, 1250, 1450, and 1650 Hz, was determined for two listening conditions (correlated and uncorrelated) under three types of uncertainty about the listening condition. Here, "correlated" means that the temporal envelopes of all five noise bands were the same, and "uncorrelated" means that only the band centered at 1250 Hz had a temporal envelope different from the other noise bands. In the no-uncertainty condition, all trials of a block consisted of correlated, or uncorrelated, waveforms. In the random-by-trial condition, correlated and uncorrelated trials were interleaved at random. In the random-by-observation-interval condition, the nonsignal interval of each trial contained the same type of waveform as the signal interval only by chance. Levels of detectability, and thus the magnitude of the comodulation masking release (CMR), were essentially the same in the first two conditions. However, in the third uncertainty condition, the CMR was practically abolished, owing almost entirely to a 6–8-dB decrease in detectability with the correlated waveforms. These results suggest a contribution of cognitive listening strategy to signal detection in comodulation masking release conditions. [Work supported by NINCDS Grant NS15895.]

SS18. Speech masking. II. Simultaneous masking thresholds under "naturalistic" listening conditions. Murray Spiegel (Bell Communications Research, 2E-252, Box 1910, 445 South Street, Morristown, NJ 07960)

This study analyzed the role of listening conditions in determining thresholds for probe tones masked by natural speech. These thresholds are of interest because they are a sensitive probe of the activity profile, or "internal" spectrum, of sounds such as speech in the auditory system. Most human performance tests are carried out under highly artificial listening conditions, which may not reflect how people listen to speech in common listening environments. In this study, a reference condition (similar to minimal uncertainty listening conditions used in many performance tests) was compared to a "naturalistic" listening condition and another, intermediate, condition. In the naturalistic listening condition, listeners did not know the frequency or the position of probe tones; additionally, they were required to attend to the semantic content of full sentences. In the reference condition, listeners knew the frequency and position of probe tones masked by single syllables. Average thresholds differed in the two conditions by 4 dB, and thresholds tended to be elevated more for higher-frequency probe tones. The results provide previously unknown information about the resolution of speech sounds in the auditory system during speech comprehension.

SS19. Temporal effects in vowel and consonant–vowel masking. Patricia E. Blake (Division of Hearing and Speech Sciences, Vanderbilt University School of Medicine, Nashville, TN 37232) and Sid P. Bacon (Department of Speech and Hearing Sciences, Arizona State University, Tempe, AZ 85287-0102)

Masking patterns were determined using 300-ms synthetic vowel (/u/, /a/, /i/) and consonant–vowel (/bi/, /gi/) stimuli as maskers in a two-interval forced-choice experiment. Thresholds were obtained for 10-ms sinusoids presented at 0- or 145-ms delays relative to masker onset for the vowel (V) masker, and at 0-, 40-, or 145-ms delays for the consonant–vowel (CV) masker. Signal frequencies ranged from 250–3600 Hz. In general, the masking patterns reflected the shape of the masker's short-term spectrum computed at the time of signal presentation. The 0-ms signal delay masking patterns for /bi/ and /gi/ clearly revealed the second-formant differences in these two stimuli. The 40-ms delay masking patterns were similarly different, though less so, even though 40 ms corresponded to the onset of the steady-state portion of the vowel (and thus at this delay the *stimuli* were identical). Formants were resolved the best for both the V and CV maskers with the 145-ms signal delay, suggesting that the auditory system requires a certain amount of time to represent most accurately the acoustic spectrum of the masker. These temporal effects are probably related to the temporal effects observed previously with pure-tone maskers. [Work supported by NIH.]

SS20. Low commonality between tests of auditory discrimination and of speech perception. Blas Espinoza-Varas and Charles S. Watson (Speech and Hearing Sciences, Indiana University, Bloomington, IN 47401)

The relation between measures of speech processing and of fine auditory discrimination abilities was investigated by means of factor analyses of the performance of normal listeners on the Test of Basic Auditory Capabilities, TBAC [Watson *et al.*, J. Acoust. Soc. Am. Suppl. 1 **71**, S72 (1982)]. The eight subtests of the TBAC include syllable identification in noise and discrimination of (a) changes in frequency, intensity, and duration in 1.0-kHz sinusoids; (b) "jitter" within a pulse train; (c) temporal order for sinusoids and nonsense syllables; and (d) detection of single-component changes in word-length tonal patterns. Data were collected in a field ($n = 127$) and using earphones ($n = 119$). Test reliability was high in both presentation conditions. Average performance was slightly higher with earphones for discrimination of intensity, of tonal patterns, and in the two speech subtests. Both groups showed large ranges of individual performance on all subtests. Factor analyses revealed similar structures for both groups, consisting essentially of two factors loaded on the nonspeech tests (complex sound/pitch and duration/intensity) and a third that is mainly a speech factor. This result, and the associated intertest correlations, points towards only a very weak relation between listeners' measured abilities to hear out fine details of acoustic waveforms and their skill at identifying speech sounds. [Work supported by NIH and AFOSR.]

SS21. The content of "timbre." Otoichi Kitamura (Department of Music, Osaka University of Arts, Kanan, Minami-kawachi, Osaka, 585 Japan)

The definition of timbre has been established by standards such as the American National Standards. According to the results of factor analytical research on sound color, the content of sound color can be well expressed by such descriptive adjectives as soft, rich, clear, etc. This aspect of sound color can be called the "impression expressed by a descriptive adjective of sound color." Furthermore, there exists another aspect of sound that identifies the sound source such as a piano, violin, etc. Generally speaking, sound impressions are considered to occur in the following order: (1) identification of the sound source, followed by recognition of detailed sound properties such as (2) the impression expressed by the descriptive adjective of the sound color, (3) loudness, (4) pitch, and (5) sound duration. The two aspects of sound, (1) and (2), are usually recognized as sound color or timbre. Accordingly, it is desirable that the content of timbre should be understood more widely as mentioned above.

SS22. Discrimination of missing fundamental frequency and the influence of competing pitch and timbre cues. Punita G. Singh (Central Institute for the Deaf, St. Louis, MO 63110)

The DL for frequency discrimination of complex tones has generally been shown to be smaller than for pure tones of equivalent $F0$. A notable exception is the case where "residue" tones are used [A. Faulkner, J. Acoust. Soc. Am. **78**, 1993–2004 (1985)]. It has been argued that the decline in performance is due to differences in timbre that make pitch discrimination difficult. The present experiment atttempts to determine such perceptual correlates of changes in fundamental frequency for complex tones presented in two-tone sequences. All tones comprised four equal-amplitude harmonics m, $(m+1)$, $(m+2)$, and $(m+3)$, with $m = 1, 2, 3, 4$, or 5. The first tone in a sequence had $F0 = 200$ Hz, while the second had $F0 = (200 + 2n)$ Hz, where $n = 0, 1, 2, 4, 8, 16$, or 32. The spectral locus of tones in a pair was either the same (m,m) or different $[(2,m)$ or $(m,2)]$, where m and 2 denote the harmonic number of the lowest component. Listeners were required to indicate if the tones in a pair were: (1) the same, (2) different in pitch, (3) same in pitch, but different in "something else," or (4) different both in pitch and "something else." Results indicate that listeners use labels 1 and 2 for sequences in which tones had the same locus, with relative use of labels systematically varying across differing magnitudes of $F0$ change. When spectral locus changes, however, a considerable variation of responses is seen, dependent on overall spectral proximity as well as amount of change in $F0$. The order of locus change across tones, whether higher or lower in frequency, is another factor that influenced label choice. [Work supported by AFOSR.]

SS23. Consonance of tones of changing frequency. Kengo Ohgushi, Shuji Morishita, Tadasu Hatoh (Department of Music, Kyoto City University of Arts, 13-6 Kutsukake-cho, Oh'e-Nishikyo-ku, Kyoto, 610-11 Japan), and Koh'ichi Kurozumi (NHK Technical and Science Research Laboratories, 1-10-11 Kinuta, Setagaya-ku, Tokyo, 157 Japan)

The tonal consonance of two sinusoids with a temporal change in frequency was measured in psychological experiments. The stimulus consisted of a pair of tone bursts presented simultaneously. The tone bursts were 1s in duration with a continuous change of frequency and had a fixed frequency ratio. The initial frequency of the lower tone from which the frequency change started was 400 Hz, and the rate of the frequency change was 1 oct/s. The ratios of the instantaneous frequencies were from 1.0–2.0. Twenty-one types of stimuli were synthesized by the computer. In the experiment, two stimuli were presented in succession to the subjects who were instructed to judge which of the two stimuli was more consonant. The results revealed the following. (1) The tonal consonance was at its minimum when the frequency ratio was around 1.05. (2) The tonal consonance increased with the increase in the frequency ratio up to around 1.2. (3) The tonal consonance kept nearly the same value and the simple frequency ratios did not result in singular points on the consonance curve when the frequency ratio was over about 1.2.

SS24. A new pitch paradox. Diana Deutsch (Department of Psychology, University of California, San Diego, La Jolla, CA 92093)

A pattern of tones is described and explored that has the following paradoxical property: When transposed by a half-octave, the identical pitches may be heard, but as reversed in time. This perceived temporal reversal may even be accomplished when the pattern is transposed by changing the speed of playback on a tape recorder. The behavior of this phenomenon under detailed parametric manipulation is described, and its implications for theories of pitch perception are discussed.

SS25. Continuous multidimensional assessment of musical performance. Seiichiro Namba and Sonoko Kuwano (College of General Education, Osaka University, Toyonaka, 560 Japan)

This study introduces a new method for the assessment of the emotional aspect of music developed by the present authors and termed the "method of continuous judgment of selected description." This is a method for the multidimensional assessment of a series of instantaneous impressions of a musical performance. Performances of "Pictures at an Exhibition," composed by Moussorgsky, were used as stimuli. In presenting the results of this experiment, two main points will be discussed. The first is the relation between instantaneous impressions during performances and the overall impression when they come to an end. The second is a comparison of the structure of the music (from an analysis of the performances using the score) with the series of impressions that the subjects registered.

SS26. Infant thresholds for short duration tone bursts. Lynne Werner Olsho and G. Cameron Marean (Department of Otolaryngology, RL-30, University of Washington, Seattle, WA 98195)

Thresholds for short duration tone bursts were measured for three- and six-month-old infants and for adults. An experimenter, blind to trial type, observed the infant's behavior. On each trial, the experimenter judged whether or not a tone had been presented and received feedback. The stimuli were 1000-Hz pure tones, with 16-ms rise/fall times and no steady-state duration. Eight tone bursts were presented at a rate of 2/s during a signal trial. Stimuli were presented monaurally via insert earphone. Thresholds were estimated using an adaptive procedure. Adults were tested using the same stimuli and procedure. Thresholds for six-month-olds averaged about 13 dB higher, while those of three-month-olds averaged about 24 dB higher than those of the adults. The size of the infant–adult difference for these short duration tones is nearly the same as that reported for longer duration stimuli [Olsho *et al.*, J. Acoust. Soc. Am. Suppl. 1 **80**, S123 (1986)]. That the effects of duration on detection threshold are quite similar for infants and adults suggests that temporal summation is mature by three months of age in humans. [Work supported by NIH.]

SS27. Resonance frequencies of the human skull *in vivo*. Anders Brandt, Peder Carlsson, Bo Håkansson (Department of Applied Electronics, Chalmers University of Technology, Göteborg, Sweden), and Anders Tjellström (ENT Department, Sahlgren Hospital, University of Göteborg, Göteborg, Sweden)

In an investigation on patients with skin penetrating titanium implants, the free-resonance frequencies of the human skull have been determined in seven subjects *in vivo*. Frequencies ranging from 100 Hz–7.5 kHz were investigated and a total of 13 to 17 resonances were identified for each subject. From the point and transfer characteristics, the lowest free-skull resonance was estimated to 990 Hz (mean) with a standard deviation of 120 Hz. Relative damping factors were found to be between 2%–15%. Intersubject variation was large, probably due to individual variations in skull geometry and mechanical properties. It is clear from both the point and transfer characteristics that the free-skull resonances have little influence on the vibrational levels of the skull, and thus should have little influence on bone conducted sound. In the transfer characteristics between the mastoids, however, narrow dips (antiresonances) were found, more than 20 dB below the average level. These antiresonances may affect bone conducted sound transmission to the contralateral ear. [Work supported by P&B Sound System Inc., Göteborg, Sweden.]

Session TT. Speech Communication IX: Special Focus Session II B, Measurement and Modeling of Speech Articulations and Articulators

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Chairman's Introduction—8:00

Invited Papers

This focus session is a continuation from Session NN. Posters from Session NN will be on display from 8:00 a.m.–12:00 noon.

8:05

TT1. Recent trends in articulatory studies in Japan. Shigeru Kiritani (Research Institute of Logopedics and Phoniatrics, University of Tokyo, 7-3-1 Hongo, Bunkyo-ku, Tokyo, 113 Japan)

Ultrasonic measurement, which at present appears to be the most convenient method for recording tongue contour, lacks an effective means of visualizing the hard palate. Techniques have been developed both at the Research Institute of Logopedics and Phoniatrics and at NTT Research Laboratories to mark position of the cranium by monitoring the relative position of the head and an ultrasonic probe. Other techniques employing magnetic or optoelectric devices are also being used for recording articulatory movements. It is expected that a new high-performance magnetic system will enable recording of positions of several magnetic pellets simultaneously through multipoint field recording and analysis. In addition to progress in articulatory measurements, new studies are emerging on the detailed measurements of acoustic effects of several articulatory factors, such as vocal tract wall radiation, the shape of the lip opening and dental structures. A compact, high-speed digital image recording system, developed and now being used at the Research Institutes of Logopedics and Phoniatrics, is also expected to be useful for the analysis of the relationship between vocal cord vibration and the acoustic characteristics of speech output.

8:25

TT2. Techniques for transducing movements of points on articulatory structures. J. S. Perkell, M. H. Cohen, and I. Garabieta (Room 36-511, Research Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge, MA 02139)

Experience with cineradiography and the x-ray microbeam has shown that midsagittal-plane data on the movements of multiple articulators inside and outside the vocal tract can be extremely useful in studying articulatory-to-acoustic relationships and strategies for speech motor control. This paper describes two recently developed alternating magnetic field systems that produce point-movement data much like the x-ray microbeam. One system uses two external magnetic-field transmitters and small biaxial receivers that are attached to the articulators; the other uses three magnetic-field transmitters and single-axis receivers. Construction, extensive testing, and simulation of both designs have shown that each system can produce useful data and each has a specific set of advantages and disadvantages. These results are described and the systems are compared with the new x-ray microbeam system at the University of Wisconsin. Examples of data are presented to illustrate their role in the study of speech production, in relation to overall requirements for articulatory data. [Work supported by NIH.]

8:45

TT3. Imaging techniques of the tongue and vocal tract. Maureen Stone (National Institutes of Health, Department of Rehabilitation, Building 10, Room 6S235, Bethesda, MD 20892)

The four most well-known imaging techniques that have been applied to tongue and vocal tract research are x-ray, computer tomography (CT), magnetic resonance imaging (MRI), and ultrasound (US). Imaging systems provide very different articulatory information from flesh point tracking systems. They provide information about the shape and position of the entire structure being imaged, rather than a single point on that structure. In addition, the images usually can be made in multiple planes providing midsagittal, parasagittal, coronal, transverse, and oblique sections of the structure. There are, however, several limitations to these systems. CT and MRI create static images only. Ultrasound cannot image hard tissue; x ray can only be used in limited amounts. This paper will discuss the advantages and limitations of imaged data, as well as the uniqueness of the measurements made from them.

9:05

TT4. Improved articulatory models. Shinji Maeda (Département Recherche en Communication par Parole, Centre National d'Etudes des Télécommunications (CNET), 22301 Lannion, France)

After a brief historical review of articulatory models, a factor analysis of the lateral shapes of the vocal tract is described. In the analysis, the profiles are specified by the sum of a small number of linear components. Studies at CNET indicate that, by using an arbitrary factor analysis in connection with the principal component analysis, it is possible to derive a linear model in which each component can be interpreted in articulatory terms, such as jaw position, tongue–body position, etc. The statistical model, therefore, can be regarded as an articulatory model. The analysis of radiocinematographic data exhibits that the same vowel can be produced differently depending on speakers and phonetic contents, suggesting a compensatory articulation. Such a model is also useful in investigating the articulatory–acoustic relationships. For example, $F1$–$F2$ trajectories are calculated by varying a specific parameter value, while the remaining parameter values are fixed appropriate for a vowel. The results show that the different jaw positions can be compensated only by the tongue–body position for front vowels, and only by the lip aperture for back vowels, to obtain a similar $F1$–$F2$ pattern. Finally, limitations of current articulatory models are discussed.

9:25

TT5. Feature extraction of articulatory motion based on the speech production model. Katsuhiko Shirai (Department of Electrical Engineering, Waseda University, 3-4-1 Ohkubo, Shinjuku-ku, Tokyo, 160 Japan)

This paper deals with two problems that exist in the transformation from an acoustic signal to the articulatory motion. The first concerns the description of articulatory motion during the utterance of voiced sounds. The configuration of the articulatory mechanism is expressed using a rather small number of articulatory parameters, and assuming simple dynamics. It is shown that articulatory movements can be produced by the step inputs for the system, and, since the coarticulation effect is reduced in the stepwise movement of the fictitious target values for each articulatory organ, vowel discrimination can be performed more easily by using these values. The second problem concerns the production model for consonants. Based on this model, the precise vocal tract shape and the position of the noise sound sources can be estimated. The estimation of the vocal tract shape, sound source, and articulatory motion using an articulatory model is a kind of inverse problem of articulatory to acoustic transformation, and requires a nonlinear optimization techinque for its solution.

9:45

TT6. Modeling speech production using dynamic gestural structures. E. Saltzman, L. Goldstein, C. Browman, and P. Rubin (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511-6695)

In the present computational model of speech production, an utterance is represented as an organization of primitive linguistic units, *gestures*, into a larger structure, a gestural score. Each distinct gesture is linked to a particular subset of *vocal tract variables* (e.g., lip aperture and protrusion) and *model articulators* (e.g., lips and jaw), and is associated with a set of time-invariant dynamic parameters (e.g., lip aperture target, stiffness, and damping coefficients). The values of the dynamic parameters and their activation intervals are computed as part of the gestural score for a given utterance using a linguistic gestural model that includes a gesture-based dictionary of English syllables and a flexible rule interpreter for manipulating dynamic parameters and intergestural phasing. The gestural score serves as input to our *task-dynamic* model of sensorimotor coordination. In this model, the evolving configuration of the model articulators results from the gesturally and posturally specific way that driving influences generated in the tract-variable space are distributed across the associated sets of synergistic articulatory components. Coarticulation effects of various sorts are automatically produced as a function of the spatial and temporal overlap of two (or more) gestures. Significantly, explicit trajectory planning is not required and the model functions in exactly the same way during simulations of unperturbed, mechanically perturbed, and coproduced speech gestures. [Work supported by NIH NS-13617, HD-01994, and NSF BNS 8520709.]

10:05–10:15

Break

10:15–12:00

Open Discussion—R. J. Porter, Jr., Moderator

Session UU. Structural Acoustics and Vibration V and Architectural Acoustics VII: Structure-Borne Noise from Vibrating Structures

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Chairman's Introduction—8:00

Invited Papers

8:05

UU1. Elastic waves in porous and particulate solids: Resonances and dead zones. Miguel C. Junger (Cambridge Acoustical Associates, Inc., Cambridge, MA 02140)

Simple asymptotic descriptions of elastic wave motions in representative micrononhomogeneous solid media are presented with the purpose of highlighting some physical phenomena. For porous media, the crucial role of the Poisson's ratio is explored: When that ratio lies between $\frac{1}{3}$ and $\frac{1}{2}$, the medium displays a "dead zone" where it cannot sustain wave motion. The propagation of dilatational and shear waves associated respectively with rigid-body translational and rotational resonances is described. Test results illustrate energy dissipation associated with relative motion between inclusions and the viscoelastic matrix.

8:25

UU2. Structural acoustics in the presence of mean flow. David G. Crighton (D.A.M.T.P., Cambridge University, United Kingdom)

Effects of mean flow on the response and radiation of a fluid-loaded elastic structure have often been ignored as being equivalent to Doppler effects on frequency and amplitude, negligible at low Mach number. Mean flow effects are, however, generally much more significant than this, and can profoundly change the whole flow of energy into and through the structure–fluid complex. This paper will illustrate these effects in the context of the response of an elastic plate to single-frequency localized drive in the presence of uniform inviscid flow. For an infinite plate, the response typically consists of exponentially amplifying instability waves downstream of the drive and two different neutral waves upstream; the drive point admittance generally has negative real part, implying energy flow into the structure and the drive, from the fluid. Approximations for various different parameter regimes will be given to predict these features. Extensions to semi-infinite and finite structures will be outlined together with interpretations in terms of negative-energy waves, and a discussion of viscous boundary layer effects. [Work supported by ONR.]

8:45

UU3. Wave propagation on thin-walled curved elastic plates with truncations. L. B. Felsen and I. T. Lu (Department of Electrical Engineering and Computer Science/Weber Research Institute, Polytechnic University, Farmingdale, NY 11735)

Pierce [ITUAM Elastic Wave Symposium, Galway, Ireland (March 1988)] has recently described the dynamics of thin-walled uniform cylindrical shells by analogy with propagation in an infinite two-dimensional uniform anisotropic medium. This analysis is generalized here, first, by allowing for edge truncations and joints, thereby generating finite or joined cylindrically curved plates, and second, by allowing for weak deviations from circular cylindrical curvature and constant thickness. The resulting equivalent anisotropic medium models involve, respectively, terminated or joined half-spaces, and spatial inhomogeneities. The analytical tools used for the generalization under time-harmonic point force excitation include infinite angular space representations, adiabatic Fourier transforms, reduced spectra (in terms of characteristic plate waves) tied to a preferred spatial coordinate, and anisotropic ray asymptotics. These analytical constructs are given a physical interpretation by recourse to the dispersion surfaces for anisotropic waves. [Work supported by David Taylor NRDC and ONR.]

THURSDAY AM

9:05

UU4. Relationship between the stored energy density and intensity. G. Maidanik and J. Dickey (David Taylor Research Center, Bethesda, MD 20084)

A model of a complex, composed of coupled one-dimensional dynamic systems, is proposed. An analytical procedure is presented that yields the steady-state stored energy density vector $\boldsymbol{\epsilon}(\mathbf{x}) = \{\epsilon_j(x_j)\}$ and the net power flow vector $\iota(\mathbf{x}) = \{\iota_j(x_j)\}$, where $\epsilon_j(x_j)$, $\iota_j(x_j)$, and x_j are the stored energy density, the net power flow, and an observation position in the (j)th dynamic system, respectively. Under appropriate averagings and assumptions, the equation that describes the stored energy vector can be reduced to the central equation in the statistical energy analysis. On the other hand, the net power flow vector is the intensity. Using the analysis, one may argue that there could exist special situations in which some corresponding elements in $\boldsymbol{\epsilon}(\mathbf{x})$ and $\iota(\mathbf{x})$ provide substantially identical information. However, in general, the two quantities are supplemental in the investigation of the energetics of a complex. Notwithstanding that, situations may arise in which the information provided by one or the other quantity may be deemed the more significant in seeking answers to specific questions.

9:25

UU5. Effects of random heterogeneity on wave fields. John J. McCoy (Department of Civil Engineering, CUA, Pangborn Hall G-10, Washington, DC 20064)

A theory of wave propagation in random media must actually refer to a suite of predictive models of the effects of a random heterogeneity. This is because a single general formalism, suitable for all stochastic experiments, would be too complicated for realistic scenarios to have any computational capability. Thus predictive models expressed in partial descriptors of a propagating radiation field, intended for specific experiments are acceptable. In this talk, a number of thought experiments for the propagation of waves in random media will be presented. The point of this is to highlight a richness in the observations one might expect to obtain as a result of random heterogeneity, and to provide a discussion of the possible physical bases for these observations. Appropriate descriptors of the radiation field for each of the thought experiments will be considered, as will be the derivation of prediction models expressed in these measures. A number of specific issues to be addressed are: the possibility of a spatial localization as a result of random heterogeneity; the physical interpretation to be given the completely coherent field; the utility of effective modulus theories; a statistical interpretation for mixture theories; energy flow and coherence formulations; etc.

9:45

UU6. Vibration and acoustic radiation from inhomogeneous structures: Effect of motion of forcing function. Mauro Pierucci (Department of Aerospace Engineering and Engineering Mechanics, San Diego State University, San Diego, CA 92182)

Homogeneous structures are known to radiate only for supersonic structural wavenumbers. The presence of inhomogeneities such as boundaries, discrete stiffeners, and variable stiffness has been shown to introduce additional modes that will radiate if the structural wavenumbers are supersonic. The radiation pattern for an infinite plate with evenly spaced stiffness gives rise to a stop–passband spectrum. The plate with a continuously varying stiffness also gives rise to similar conditions. All of the previous analyses have always considered stationary forcing functions. In this paper, the analyses for the fixed forcing function will be reviewed and then they will be generalized to the case of the forcing function moving along the plate at speeds up to and greater than the wave speeds in the solid. The results presented will show that for certain values of the frequency, rib spacings, and forcing function velocity, traveling wave modes may change direction, decaying modes may become traveling modes, and modes may change from subsonic to supersonic and vice versa depending upon the relative motion of the forcing function. The effect of the motion may thus enhance or reduce the acoustic radiation.

10:05

UU7. Zeros of a transfer function in a structural vibrating system. Mikio Tohyama (NTT Human Interface Laboratories, 3-9-11 Midoricho, Musashino, 180 Japan) and Richard H. Lyon (Department of Mechanical Engineering, Massachusetts Institute of Technology, Cambridge, MA 02139)

This paper analyzes a transfer function (TF) of a structural vibrating system in the complex-frequency domain based on wave theory. There are three cases of zeros in a TF. One is the usual case where a zero is located on the pole line connecting two adjacent poles. Another is the double-zero case, that is, there are two zeros on the pole line between two adjacent poles. These zeros mentioned above are minimum-phase zeros. The third case, however, can contain a nonminimum-phase zero. A pair of minimum- and nonminimum-phase zeros is located symmetrically with respect to each other at equal distances from the pole line. One member of this pair of zeros can be a nonminimum-phase zero under the condition that the damping is small. Such zeros have been numerically confirmed using the TF of three-dimensional space vibration. The truncation effects of an impulse response on the zeros are also mentioned.

10:25

UU8. Acoustic radiation from line excited plates—Asymptotic series. Hong-Yuan Hsu (Department of Mechanical Engineering, University of Kentucky, Lexington, KY 40506) and Sabih I. Hayek (Department of Engineering Science and Mechanics and the Applied Research Laboratory, The Pennsylvania State University, University Park, PA 16802)

The theoretical analysis for predicting the acoustic radiation from an infinite fluid-loaded, elastic plate excited by a line force is presented. The acoustic pressure radiated by this coupled plate–fluid system is given in an integral representation. A uniformly asymptotic series solution, obtained by the use of the modified saddle point and subtraction of poles method, is valid in the farfield and the nearfield, and is especially accurate at the coincidence angle where the first-order approximation by the saddle point method overpredicts the acoustic pressure. The uniformly asymptotic series solution also predicts a decaying but nonvanishing surface acoustic pressure while the approximate solution from the saddle point method vanishes at the grazing angle. The solution was also obtained by numerical integration along the steepest descent path. Results obtained from the numerical integration are compared with the uniformly asymptotic series solution and excellent agreement has been observed.

10:37

UU9. Analysis of vibration transmission through periodic structures using a power flow approach. J. M. Cuschieri (Center for Acoustics and Vibrations, Department of Ocean Engineering, Florida Atlantic University, Boca Raton, FL 33431)

The analysis of the parameters controlling the transmission of vibrational power through coupled structures can be performed using a structural power flow approach. In this paper, a periodic structure is considered with equal length spans between the stiffeners or the supports. With the power flow technique, the global structure is divided into subelements for which structural mobility functions can be defined. Also, structural mobility elements can be introduced at the interfaces between the subelements to represent different types of supports or stiffeners or some other interface characteristic. Thus, using this approach, the influence of such parameters as translational and rotational inertia, stiffness, and damping of the supports or stiffeners, and the stiffness, damping, and other structural characteristics of the separate spans, can be determined using this technique. This approach is applied here to a thin periodic beam with different types of stiffeners and supports, and some of the analysis results are presented. [Work supported by NASA.]

10:49

UU10. Power flow finite element analysis of beam networks. Donald J. Nefske and Shung H. Sung (Engineering Mechanics Department, General Motors Research Laboratories, Warren, MI 48090-9057)

A power flow analysis was previously developed for predicting the vibration response of structural and acoustic systems to high frequencies at which the traditional finite element method is no longer practical [D. J. Nefske and S. H. Sung, ASME NCA 3, 47–54 (1987)]. As compared to statistical energy analysis, which predicts an overall vibration response of each individual subsystem, the power flow analysis enables one to predict the spatial variation of the vibration response within each subsystem, as well as the power flow and vibration response throughout the entire system. The present paper considers application of the power flow method to built-up frame structures in the form of beam networks. The power flow analysis considers excitation of multiple wave types in the individual beams and utilizes transmission efficiencies to couple the wavetypes at the beam junctions. The power flow solution is obtained by modifying the properties of a standard structural finite element model to form a power flow finite element model, which is then solved using MSC/NASTRAN. The predictions of the power flow finite element method are compared with the measured vibration on an actual structure and with the predictions of the traditional finite element method.

11:01

UU11. Axisymmetric vibrations of an elastic fluid-loaded cylinder. Douglas Photiadis (DTRC, Code 1965, Bethesda, MD 20084-5000)

The response of an infinite elastic cylindrical shell immersed in an acoustic medium is examined analytically. The relatively low frequency range, $(ka) < 3$–4, is investigated. The cylinder is subject to a localized axisymmetric excitation, a ring drive. Starting from the known integral expression for the Green's function, approximation techniques were applied to extract the dominant behavior of the surface field. Expressions for the response are given for distances greater than an acoustic wavelength from the source location.

11:13

UU12. Gaussian beam propagation on cylindrical thin shells. A. N. Norris (Department of Mechanics and Materials Science, Rutgers University, Piscataway, NJ 08855-0909)

The flow of acoustic energy on cylindrical thin shells can be understood by looking at how disturbances in the form of Gaussian beams propagate. Gaussian beams are also useful tools for considering radiation from a given source, if the source can be easily represented as the sum of a few distinct beams. The recent work of Pierce [A. D. Pierce, *Proceedings of IUTAM Symposium on Elastic Waves*, Ireland, 1988 (to be published)] showed that the equations are very similar to those for an unbounded two-dimensional anisotropic and dispersive medium. Using this analogy and a recently developed formalism for Gaussian wave packets in anisotropic solids [A. N. Norris, Wave Motion 9, 509–532 (1987)], the equation governing the change in wave front curvature is derived. Gaussian beams are obtained by taking the curvature to be complex valued initially. The change in curvature then determines the narrowing or broadening of the beam. The anisotropy causes the broadening to occur in a direction that is not perpendicular to the direction of propagation. The strong dispersion of the flexural wave introduces effects not previously seen in Gaussian beams. The influence of fluid loading will also be mentioned.

11:25

UU13. Experimental identification of a noise prediction model of a vibrating structure using optimal estimation techniques. R. Aquilina, J. C. Flandrin, and E. Gaud (Centre d'études et de recherches pour la discrétion acoustique des navires, DCAN Toulon, 83 800 Toulon Naval, France)

A novel statistical method of estimating a noise prediction model of a vibrating structure has been developed using optimal estimation techniques. The aim of this method is to derive the optimal transfer functions between a set of vibratory information taken on the structure, and the radiated noise. An underlying problem to be solved is then the selection of the minimal acceleration subset necessary to "explain" this radiated noise. The statistical approach lies in the achievement of an important number of experimental excitations of the structure and the measurements of the frequency response functions (FRF) of the model inputs (accelerations) and the output (radiated noise). Each noisy measurement is then regarded as a random variable, and the optimal prediction model is obtained by minimizing the variance of residual errors after the selection of the more significant inputs using partial correlation analysis. The measurement noise characteristics are derived from the coherence function information relative to each FRF measurement. Very good experiment results are presented concerning a submersed vibrating shell over a wide frequency range.

Session VV. Underwater Acoustics VI: Shallow Water Acoustics I

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Chairman's Introduction—8:00

Invited Papers

8:05

VV1. Shallow water acoustics on the Canadian East Coast Continental Shelf. Philip R. Staal, Steven J. Hughes, Dale D. Ellis, and David M. F. Chapman (Defense Research Establishment Atlantic, P.O. Box 1012, Dartmouth, Nova Scotia B2Y 3Z7, Canada)

The Canadian East Coast Continental Shelf is a large area (roughly 600 000 km^2) of complex bottom types geologically related to the glacial cover and low water level of the recent Ice Age. The bottom types include thickly sedimented sand banks under less than 100 m of water, thickly sedimented silt/clay basins under more than 200 m of water, thinly covered or bare glacial till, and outcropped rock. This paper describes DREA measurements of underwater acoustic propagation and ambient noise in the 1- to 1000-Hz frequency band, and attempts to explain these measurements. Acoustic propagation over thickly sedimented seabeds can be explained by models that assume "fluid" sediment layers (i.e., no shear waves). However, over thinly sedimented rocky seabeds the acoustic models need to include shear waves to explain the measured acoustic propagation, in particular the high losses between 10 and 100 Hz. Ambient noise is affected by the seabed, mainly by the difference in acoustic propagation over unlike seabed types, from distant sources.

8:25

VV2. Developments in shallow water seismoacoustics. T. G. Muir and T. Akal (SACLANT Undersea Research Centre, Viale San Bartolomeo 400, 19026 La Spezia, Italy)

Since the 1970s, when low-frequency acoustics in shallow water became a priority topic, developments have progressed from propagation loss in the frequency domain, to waveguide phenomena dominated by bottom interaction, to ocean bottom seismology and studies of interface waves involving sediment rigidity. Emphasis has now been placed on the entire propagation medium, including the water column, the sediment interface, the volume of unconsolidated sediments, and the underlying bedrock strata; as well as on the entire spectrum of wave types, which in reality are part and parcel of a single unified process. Sophisticated research tools have been developed for these studies, including remotely operated, three-axis seismometers, machines to deploy them at depth within the sediments, and various computer models, such as the SAFARI code that can address the creation and propagation of seismoacoustic waves. Results obtained with these tools in research done in the shallow seas surrounding Italy, Norway, and Spain are presented.

8:45

VV3. An estimation method of vertical sound-speed profiles in shallow water using multipath propagation. Hiroyuki Hachiya, Shigeo Ohtsuki, and Motoyoshi Okujima (Research Laboratory of Precision Machinery and Electronics, Tokyo Institute of Technology, Nagatsuta, Midori-ku, Yokohama, 227 Japan)

To observe changes in shallow water structure, a method of estimating the vertical sound-speed profile in shallow water is presented. "Shallow" means a water depth of less than about 1500 m in which sound can be propagated a certain distance by repeated reflections at both surface and bottom. In such water, the acoustic characteristics of both the surface and the bottom are important determinants of the sound field. The acoustical characteristics of shallow water in an acoustic transmission experiment that was conducted in Sagami Bay using a 5-kHz pulsed cw signal at a range of 10 km are examined. According to the results, bottom-and/or

surface-reflected rays are stable in time and have identifiable ray paths. Using a shallow water model, a simulation to estimate the vertical sound-speed profile not from the travel times of the rays but from the travel time differences between the bottom- and/or surface-reflected rays was carried out. Travel time differences between individual rays could easily be measured by a clock, and estimated results agree well with given correct models.

9:05

VV4. Horizontal refraction of normal modes due to variation in bathymetry and subbottom sedimental structure. Kazuhiko Ohta (5th Research Center, Technical Research and Development Institute, Japan Defense Agency, 3-13-1 Nagase, Yokosuka, 239 Japan)

The horizontal refraction of normal modes propagating over a sloping bottom with range-dependent sedimental structure is studied within the context of adiabatic approximation. Numerical examples show that the effect of the rangewise variation of the subbottom sedimental structure on the horizontal refraction of modes is even more significant than that of the bathymetric variation, depending on the frequency and the mode considered. To investigate the relation between the bottom interaction and the horizontal refraction of modes, the range evolution of the local eigenvalue along the slope, which is proportional to the horizontal refraction index, is examined. Its derivative with respect to range has been derived from the depth equation and can be expressed in an analytical form using depth gradients, mode-depth functions, and the differences in both vertical wavenumbers and densities at each interface. This result may explain the behavior of the horizontal refraction of modes.

9:25

VV5. Bottom sediments—Viscoelastic, poroelastic, or poro-viscoelastic? Robert D. Stoll (Lamont–Doherty Geological Observatory, Palisades, NY 10964)

The velocity and attenuation of seismic waves are known to be frequency dependent in virtually all marine sediments. The frequency dependence results from various kinds of viscous losses in the pore water as well as from intergranular friction in the skeletal frame. The viscous losses fall into two broad categories—losses due to overall motion of the fluid field relative to the skeletal frame and local fluid motion (near the grain contacts) and in cracks and crevices. The cumulative effect of these kinds of energy loss are generally described by a phenomenological model (i.e., viscoelastic, poroelastic, poro-viscoelastic, etc.) with input parameters that are related to measurable physical properties of the sediment. Practical geoacoustic models may range from a simple viscoelastic model with a single relaxation time, valid over a narrow frequency band, to very complex models based on the more general Biot theory that require many more input parameters. In this paper some of the many factors that influence the choice of a practical model are discussed and illustrations of both "simple" and complex cases are given. [Work sponsored by ONR, code 11250A.]

9:45

VV6. Acoustic propagation in inhomogeneous underwater acoustic waveguides. Suzanne T. McDaniel (Applied Research Laboratory, The Pennsylvania State University, University Park, PA 16804)

The propagation of sound in horizontally stratified media may be treated using a variety of techniques. Realistic shallow-water environments may, however, exhibit significant departures from this ideal geometry. This paper reviews various methods, both deterministic and stochastic, for handling range-dependent shallow-water propagation. Stochastic coupled-mode theory is applied to consider the effects of sea surface scattering. It is shown that for low enough frequencies, adiabatic normal mode theory provides a sufficient treatment, in accord with the findings of both laboratory and at sea measurements. This same technique is also applied to consider the effects of inhomogeneities within the seabed and water column on the propagated field and its spatial coherence. Implications for matched field processing are discussed, and methods of estimating performance degradation due to rough boundary interaction are presented.

10:05

VV7. The performance of an eigenvalue-based stabilization algorithm for high-resolution beamformers: Experimental tests with shallow water data. Robert F. Gragg and Bruce H. Pasewark (Naval Research Laboratory, Code 5160, Washington, DC 20375)

This work investigates the performance of high-resolution beamforming techniques as used with acoustic arrays in shallow water. The maximum likelihood method (MLM) is the high-resolution technique of principal interest and the maximum entropy method (MEM) is addressed briefly. The topics dealt with are the performance of beamformers with computer-simulated data, the performance degradation caused by system phase errors, and performance with actual field data. High resolution methods are found to suffer performance degradation in the form of instabilities in the beamform outputs. Results indicate that the proper pre-processing method (namely, the elimination of the lowest-lying eigenvalues from the cross-spectral density matrix) can stabilize maximum likelihood beamformers so that they outperform conventional beamformers at the tasks of resolving targets with similar bearings and tracking them through bearing crossings. No stabilizing method was found for maximum entropy beamformers. [Work supported by ONT.]

10:25

VV8. A hydrodynamic siren as an underwater sound source. Shigeru Yoshikawa (5th Research Center, Technical R&D Institute, Japan Defense Agency, Nagase, Yokosuka, 239 Japan) and Kiyoshi Koyano (Totsuka Works, Hitachi Ltd., Totsuka, Yokohama, 244 Japan)

This paper describes the acoustic characteristics of a hydrodynamic siren in which a perforated disk (rotor) rotates in front of a similar disk (stator) fixed with very slight clearance and compressed water passes through holes in the rotor and stator. The stream is interrupted with the frequency given by the product of the rotation rate times the hole number. The rotor and stator have 8 and 20 holes on their inner and outer circles, respectively. The sound due to the greatest common divisor of 8 and 20 (i.e., 4) is also generated by the superposition of two waveforms derived from 8 and 20 holes. The siren thus produces thickly populated line spectrums that extend from 0.1–5 kHz. The motor and pump are controlled by inverters to stably adjust the rotation rate of the rotor and the pressure of the water, respectively. The maximum sound pressure level integrated over the above spectrum range reaches about 190 dB *re*: 1 Pa at 1 m when the water pressure is 9 kg/cm^2. This siren has been used as a jammer or broadband mixed sound projector for sea experiments on adaptive beamformers.

Contributed Papers

10:45

VV9. Chaos and hyperchaos in shallow water acoustics. Frederick D. Tappert, Gustavo J. Goni, and Michael J. Brown (Applied Marine Physics Division, RSMAS, University of Miami, 4600 Rickenbacker Causeway, Miami, FL 33149)

The ray equation for shallow water sound propagation in a "topless" ocean over a sinusoidal bottom can, under certain circumstances, be reduced to the area preserving standard mapping [F. D. Tappert and M. G. Brown, J. Acoust. Soc. Am. Suppl. 1 **83**, S37 (1988)]. For ray paths interacting with a flat sea surface and the ocean bottom, the Poincaré sections and the Lyapunov exponents evidence the chaotic behavior of the mapping. Numerical results show the sensitivity of the equations to changes in the environmental and initial conditions. The mapping dependence on the stochasticity parameter K (environment) and on the initial conditions is studied. For any nonzero value of K it can be shown that there exists a chaotic behavior (hyperchaos), in contrast to the critical value of K, $K_c \cong 0.97$, for a "topless" ocean, under which no chaotic behavior is observed.

10:57

VV10. Generation of acoustic waves by an impulsive point source in a fluid/porous-medium configuration with a plane boundary. Sÿtze M. de Vries and Adrianus T. de Hoop (Laboratory of Electromagnetic Research, Faculty of Electrical Engineering, Delft University of Technology, P. O. Box 5031, 2600 GA Delft, The Netherlands)

The space-time acoustic wave motion generated by an impulsive monopole point source in a fluid/porous-medium configuration with a plane boundary is calculated with the aid of the modified Cagniard technique. To describe the macroscopic propagation of linear acoustic waves through the fluid-saturated porous solid, four coupled partial differential equations are used that follow upon applying a volume averaging procedure to the two constituents, a perfectly elastic solid and an ideal fluid of the porous medium. The source is located in the fluid part of the configuration. Numerical results are presented for the reflected-wave acoustic pressure, especially in those regions of space where head waves occur. There is a marked difference in time response in the different regimes that exist for the wave speed in the fluid in relation to the different wave speeds in the porous medium (fast compressional, slow compressional, shear). These differences are of importance to the situation where the reflected wave in the fluid is used to determine experimentally the acoustic properties of the porous medium.

11:09

VV11. Pulse response measurements over three 45-km propagation paths in the Florida Straits. Harry A. DeFerrari and Charles Monjo (Department of Applied Marine Physics, RSMAS, University of Miami, Miami, FL 33149)

Three point reciprocal transmission experiments, undertaken in the Florida Straits, have yielded a 35-day time history of shallow water pulse responses for 45-km ranges in a depth of approximately 550 m. The transmitted pulses consist of 4 cycles of a 460-Hz carrier. Several distinct and separable arrivals persist throughout the 35-day period for all three ranges. The sound-speed profile has a 90-m surface duct above a negative gradient of approximately 0.15 (1/s). Both the sound speed and bathymetry have complicated range dependence and the medium is highly variable in time owing to meanders of the core of the Florida Current. Yet, the arrival patterns are remarkably stable. The observed time spread of all arrivals is nearly a full second. Travel times of the earliest arrivals are consistent with surface duct transmission, whereas later arrivals are consistent with ray model predictions for refracted bottom reflected (RBR) rays. The RBR arrivals are spread over approximately 0.3 s. Ray models, normal mode, and PE models are used to predict and identify intermediate arrivals. [Work supported by ONR.]

11:21

VV12. A comparison between broadband and narrow-band measurements of bottom properties in shallow water in the Gulf of Mexico. James F. Lynch, George V. Frisk, and Subramaniam D. Rajan (Department of Ocean Engineering, Woods Hole Oceanographic Institution, Woods Hole, MA 02543)

In order to assess the relative merits of two different types of acoustic measurements in determining bottom geoacoustic properties, both narrow-band and broadband data taken at the same site in the Gulf of Mexico near Corpus Christi, TX have been inverted. The narrow-band data, taken by Lynch and Frisk, were obtained over a well-sampled 5-km track at 50 and 140 Hz. These data were then processed using the Hankel transform to display the modal eigenvalue structure of the waveguide at those frequencies [G. V. Frisk and J. F. Lynch, J. Acoust. Soc. Am. **76**, 205–216 (1984)]. The broadband data were taken by L. A. Rubano [J. Acoust. Soc. Am. **67**, 1608–1613 (1980)] in a band between 20 and 250 Hz and yielded a group velocity dispersion curve for the first normal mode. Both types of data were then inverted using perturbative inverse techniques developed by Rajan *et al.* [J. Acoust. Soc. Am. **82**, 998–1017 (1987)], producing both bottom geoacoustic models and error estimates (resolution and variance) for the bottom models. Comparisons of the geoacoustic models with each other, as well as with a third model generated by a NORDA group using a compilation of archival records [Matthews *et al.* NORDA Report 120 (June 1985)] are made. [Work supported by ONR.]

11:33

VV13. Shallow water propagation under the Arctic ice canopy. David Wegmann and Herman Medwin (Physics Department, Naval Postgraduate School, Monterey, CA 93943)

Propagation in a shallow water waveguide, covered by a layer of "ice," is being studied in a 10-m-long laboratory waveguide that models both the scale and the physical properties of the Arctic. Both smooth ice and rough ice effects are investigated. Comparisons are made with predictions of the Naval Research Laboratory computer model, KRAKEN. Although the ice canopy thickness is only a small fraction of a wavelength, the sound field is distinguishable from the usual Pekeris waveguide behavior by evidence of the presence of solid-borne modes in the ice. The observed effects are related to the stimulation of ice head waves, which has been reported previously for deep water regions [J. Acoust. Soc. Am. **83**, 1794 (1988) and J. Acoust. Soc. Am. Suppl. 1 **82**, S31 (1987)]. [Work supported by the office of Naval Research.]

11:45

VV14. Propagation loss at Arctic under-ice cover. E. Y. T. Kuo (U.S. Naval Underwater Systems Center (NUSC), New London, CT 06320)

The propagation loss of low-frequency acoustic energy at Arctic under-ice cover depends on various parameters. This talk gives the combined effects of (1) the under-ice and over-ice roughnesses; (2) the ice layer thickness; (3) the shear wave speed and the absorption represented, respectively, by the real and imaginary part of the longitudinal plate wave speed; and (4) the physical properties of ice and water. At low frequencies, the Arctic ice cover can be approximated by a rough thin plate. Similarly, the roughness scattering process can be approximated by a small perturbation theory [H. Marsh, J. Acoust. Soc. Am. **33**, 330–333 (1961); E. Y. T. Kuo, J. Acoust. Soc. Am. **36**, 2135–2142 (1964) and **81**, 1762–1766 (1987)]. Compared to experimental propagation loss, the new results improve considerably from the previous predictions that were based on a pressure release boundary condition [F. R. DiNapoli and R. H. Mellen, NUSC TM851130 (1985); E. Y. T. Kuo, NUSC TM871118 (1986)]. It is found that the combined effect of roughness scattering and ice absorption dominates over that of the shear wave speed at grazing angles. [Work supported by IR/IED program at NUSC.]

THURSDAY MORNING, 17 NOVEMBER 1988 IAO NEEDLE/AKAKA FALLS ROOM, 9:30 A.M.

Meeting of Accredited Standards Committee S12 on Noise

to be held jointly with the

Technical Advisory Group for ISO/TC 43/SC1 Noise and ISO/TC 94/SC12 Hearing Protectors

W. Melnick, Chairman S12
Ohio State University, University Hospital Clinic, 456 Clinic Drive, Columbus, Ohio 43210

H. E. von Gierke, Chairman, Technical Advisory Group for ISO/TC 43/SC1
Biodynamics & Bioengineering Division, AFAMRL/BB U.S. Air Force, Wright-Patterson AFB, Dayton, Ohio 45433

Standards Committee S12 on Noise. Working group chairs will report on their progress under the plan for the production of noise standards. The interaction with ISO/TC 43/SC1 and ISO/TC 94/SC12 activities will be discussed. A report will be given on the meeting of ISO/TC 43/SC1, to take place in Toronto, Canada, from 11–14 October 1988.

Session WW. Speech Communication X: Perception (Poster Session)

Sadaoki Furui, Cochairman
NTT Human Interface Laboratories
3-9-11 Midori-cho
Musashino, 180 Japan

Ralph N. Ohde, Cochairman
Division of Hearing and Speech Sciences
Vanderbilt University School of Medicine
Nashville, Tennessee 37232

Contributed Papers

Posters should be set up before 1:30 p.m. All posters must be removed by 4:30 p.m. to prepare for the Banquet.

All posters will be displayed from 1:30 to 4:25 p.m. To allow contributors the opportunity to see other posters, contributors of papers WW1 to WW11 will be at their posters from 1:30 to 2:30 p.m., contributors of papers WW12 to WW22 will be at their posters from 2:30 to 3:30 p.m., and contributors of papers WW23 to WW33 will be at their posters from 3:30 to 4:25 p.m.

WW1. The development of cues to the perception of the [m]-[n] distinction in CV syllables. Ralph N. Ohde (Division of Hearing and Speech Sciences, Vanderbilt University School of Medicine, Nashville, TN 37232)

The contribution of the nasal murmur and vocalic formant transitions to the perception of the [m]-[n] distinction by adult listeners was investigated for speakers of different ages. Three children, ages 3, 5, and 7, and an adult female and male produced CV syllables consisting of either [m] or [n] and followed by [i,æ,u,ɑ]. Three productions of each syllable were modified according to several waveform editing techniques. Preliminary results of listening tests indicate that the murmur and vocalic transitions provide cues to place of articulation, with the latter property more prominent in perception than the former. The simultaneous presence of murmur and vocalic transition cues improved perception of place of articulation for some syllables, particularly for children's speech. The results will be discussed relative to the role of variability in production and integrated cues to the perception of place of articulation of nasal consonants in speech development. [Work supported in part by Biomedical Research Support Grant No. RR-05424.]

WW2. Auditory-perceptual analysis of selected syllables. James D. Miller (Central Institute for the Deaf, St. Louis, MO 63110)

Analyses of consonant–vowel syllables (CVs) in terms of the auditory-perceptual theory of phonetic recognition will be presented. Examples of CVs will include a voiceless stop, a voiceless fricative, a nasal, and an approximant paired with monophthongal vowels. Spectral analyses are used to locate the formant peaks and to track these during the course of the syllable, producing a sequence of spectra, one for each ms of waveform. Formant and $F0$ information from these sequences is then converted into sensory and perceptual paths in the theory's auditory-perceptual space. This space contains subspaces, called perceptual target zones. The activation of a zone results in the output of a phonetic code. While the exact conditions for this activation are not yet known, it appears that certain aspects of the behavior of a perceptual path in relation to the perceptual target zones can determine the phonetic transcription of a syllable. [Work supported by NINCDS and AFOSR.]

WW3. Evidence for the 3-Bark integration interval. Brian A. Hanson and Hector Raul Javkin (Speech Technology Laboratory, 3888 State Street, Santa Barbara, CA 93105)

It is generally accepted that sound energy within a 1-Bark interval is integrated by the ear. Chistovich *et al.* [in *Frontiers of Speech Communication Research*, edited by Lindblom and Ohman (Academic, New York, 1979), pp. 143–157] found evidence of a larger integration interval, of approximately 3 Bark, for vowel perception. Klatt [J. Acoust. Soc. Am. Suppl. 1 77, S7 (1985)] found that listeners could distinguish between vowels whose formant differences were compensated by bandwidth differences (so that cue trading did not occur) and concluded that the 3-Bark interval was not supported. The present paper analyzes the effect of different intervals on the center of gravity analysis described in Javkin *et al.* [J. Acoust. Soc. Am. Suppl. 1 **82**, S81 (1988)] adapted from Chistovich and Chernova [Speech Commun. **5**, 3–16 (1986)], and compares that analysis with a perceptual experiment testing the effect of harmonics on the perception of formants. The results support the hypothesis that the minimum interval is about 3 Bark. Our model may explain the lack of cue trading between formants and bandwidths.

WW4. Learning to identify phonemic orders. Brad S. Brubaker and Richard M. Warren (Department of Psychology, University of Wisconsin—Milwaukee, Milwaukee, WI 53201)

Several laboratories have reported that the order of items within repeating vowel sequences cannot be identified at item durations below 100 ms. This is puzzling, since phonemic sequences in speech consist of items that are considerably briefer. The present study demonstrates that listeners can, without feedback or knowledge of results, readily learn to name phonemic orders at item durations even briefer than those of normal speech. The listeners first identified different orders of components in repeating three-item vowel sequences at a few hundred ms per item (an easy task), and were then required to identify orders of these sequences at regularly decreasing item durations. By a series of successive generalizations to ever shorter items, order was named down to about two glottal pulses/vowel (the shortest duration used). Within the durational range of vowels in speech, our listeners distinguished different arrangements by their resemblances to particular words. Below this range, listeners employed qualitative differences of a nonverbal nature. Implications for theories of speech perception will be discussed. [Work supported by NIH.]

WW5. Rate-dependent perception of interpersonal characteristics. Stanley Feldstein, Faith-Anne Dohm, and Cynthia L. Crown (Department of Psychology, University of Maryland—Baltimore County, 5401 Wilkens Avenue, Baltimore, MD 21228 and Department of Psychology, Xavier University, Cincinnati, OH 45207)

The hypothesis, derived from the similarity-attraction literature, was that listeners describe speakers in more positive ways when they judge speakers' global speech rates to be similar to their own. Forty-five male and female listeners judged the global rates of three male and three female speech samples and how those rates compared with their own rates. The speaker of each sample was evaluated in terms of ten unipolar, adjective scales, each of which ranged from 0 to 9, with the higher score having the higher valence. The scores of the ten scales were then averaged to provide a total attribution valence score, and were also divided to provide "competence" and "affability" factor scores. The scores were subjected to appropriate regression analyses that included as independent variables speaker and listener gender and the perceived and actual differences between the speaker and listener rates. In support of the hypothesis, listeners assigned more positive total attribution values to those speakers whose rates were similar to their own, although their gender and the actual differences between their rates and the speakers' rates jointly influenced their evaluations of competence.

WW6. Listener experience and perception of voice quality. Jody Kreiman and Bruce R. Gerratt (V. A. Medical Center, Audiology and Speech Pathology (126), Wilshire and Sawtelle Boulevards, Los Angeles, CA 90073 and School of Medicine, UCLA, CHS-132, Los Angeles, CA 90024)

This study examines the role of listener experience with populations of voices in the perception of vocal quality. Prototype models suggest that the strategy that naive listeners use to evaluate abnormal vocal qualities will differ significantly from that of listeners experienced with dysphonic populations. Listener groups should *not* differ in perceptual strategy for normal voices, where their experience is equal. The *nature* of differences in strategy is also examined: as listeners gain experience with populations of voices, do they hear them in more complicated ways, resulting in higher dimensional multidimensional scaling solutions and/or lower $r2$ values for these solutions? Alternatively, does experience increase the efficiency with which listeners utilize a relatively constant set of perceptual parameters? Six naive and six experienced listeners judged the similarity of 17 dysphonic and 17 normal voices. Separate MDS solutions were found for each listener group for each voice set, and regressions compared the solutions. The relative complexity and efficiency of individual perceptual strategies are also discussed.

WW7. Factors affecting the integration of auditory and visual information in speech: The effect of vowel environment. Kerry P. Green, Patricia K. Kuhl (Department of Speech and Hearing Sciences, University of Washington, Seattle, WA 98195), and Andrew N. Meltzoff (Department of Psychology, University of Washington, Seattle, WA 98195)

In the McGurk effect, observers typically report the illusory syllable /da/ when they hear the auditory syllable /ba/ presented in synchrony with a video display of a talker saying /ga/. While the effect itself has been well established, there is still little research on the conditions under which the effect occurs. In the experiment reported here, the number of illusory /d/ responses to the auditory /b/–visual /g/ combination is examined in three vowel environments: /a/, /i/, and /u/. The results of this study indicate that the magnitude of the illusion is not the same across different vowel environments. It appears to be strongest for the /i/ vowel, moderate for /a/, and almost nonexistent for /u/. The results thus show that vowel environment is an important factor in determining the magnitude of the McGurk effect, which needs to be considered in accounts of auditory–visual integration during speech perception. [Work supported by NIH.]

WW8. Effects of attention on the phonetic importance of acoustic cues. Peter C. Gordon and Jennifer L. Eberhardt (Department of Psychology, Harvard University, Cambridge, MA 02138)

The effect of attentional capacity on recognizing phonetic segments was studied by having subjects identify synthetic speech sounds while simultaneously performing a visual–manual tracking task. The difficulty of the tracking task was manipulated in order to vary the amount of attention available for speech perception. Within the speech stimuli, acoustic cues to segment identity were manipulated so that trading relations could be assessed under different levels of attention. The relative importance of the acoustic cues to perceptual identification changed, depending on the difficulty of the concurrent tracking task. It appears that acoustic cues that are easily encoded make an increased contribution to phonetic judgments when listeners are unable to pay close attention to a speech sound. In this way, attention to the speech stimulus is similar to other factors (e.g., environmental interference, stage of development, and hearing loss) that also affect the relative contribution of acoustic cues to phonetic perception. It also appears that the initial stages of speech perception can make use of general, modality-independent attentional resources. [Work supported by AFOSR.]

WW9. Identification of stops in consonant sequences extracted from continuous speech. Lori F. Lamel (Room 36-545, Department of Electrical Engineering and Computer Science and Research Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge, MA 02139)

At the 114th meeting of the Acoustical Society of America, experiments on the listener's ability to identify singleton stop consonants in syllable-initial and noninitial position, and stops in noninitial homorganic nasal-stop clusters, were reported. The stimuli, consisting of portions of speech extracted from continuous sentences, were drawn from a corpus of about 3600 sentences spoken by over 450 talkers. This paper further investigates the effect of intervening consonants on the listener's decision. Ten listeners identified stops in clusters with semivowels. With the exception of /dr/ and /tr/ (which were confused primarily with the /ǰ,č/), the listener's performance (96.1% correct) was comparable to that of singleton stops (97.1%). Another group of listeners identified stops preceded by /s/ or /z/. The listener's performance (overall 88.3%) was dependent upon the identity of the fricative and whether or not the stop was in a cluster with the preceding /s/. Results will be compared with singleton stop identification tasks. [Work supported by DARPA under Contract No. N00014-82-K-0727, monitored through the Office of Naval Research.]

WW10. Fundamental frequency provides voicing information even with unambiguous VOTs. D. H. Whalen (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511), Arthur S. Abramson (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511 and Department of Linguistics, University of Connecticut, 341 Mansfield Street, Storrs, CT 06268), Leigh Lisker (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511 and Department of Linguistics, University of Pennsylvania, Philadelphia, PA 19104), and Maria Mody (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511 and Graduate Center, Department of Speech and Hearing Sciences, City University of New York, 33 West 42nd Street, New York, NY 10036)

Earlier work [A. S. Abramson and L. Lisker, in *Phonetic Linguistics* (1985)] demonstrated that falling fundamental frequency ($F0$) after a syllable-initial stop was a cue to voicelessness, and that flat or rising $F0$ was a cue for voiced stops, but only when the voice onset time (VOT) was ambiguous. The present study replicated that finding with seven VOT values and five onset $F0$ values. In the first condition, subjects identified the stop as "b" or "p." Results were nearly identical to the previous experiment. A second condition included not just the stop decision, but a reaction time as well. Here, inappropriate $F0$ slowed response time even for unambiguous VOTs. A final condition was, like the first, identification without time pressure. Here, it was found that subjects were distinguishing all five levels of $F0$ onset so that, the lower the onset was, the more "b" responses were obtained in the ambiguous region. Thus $F0$ contributes to the voicing distinction, even when the categorization is not changed. Also, $F0$ cues a "voiced" response incrementally as it starts below the $F0$ of the remainder of the syllable. [Work supported by NIH Grant No. HD-01994.]

WW11. On the role of the fundamental frequency in vowel perception. Tatsuya Hirahara (ATR Auditory and Visual Perception Research Laboratories, Twin 21 MID Tower, 2-1-61 Shiromi, Higashi-ku, Osaka, 540 Japan)

Vowel identification tests were carried out using 200 synthesized vowel-like stimuli to examine the role of the fundamental frequency $F0$ in vowel perception. These stimuli were synthetic versions of the five Japanese vowels, /i/, /e/, /a/, /o/, and /u/, of which the $F0$ and/or the formant frequencies Fi ($i = 1,2,3,4$) were modified: ten $F0$ values were formed by adding $n/3$ Bark ($n = 0,1,...,9$) to the original $F0$. Four formant frequency sets were formed by adding m Bark ($m = 0,1,2,3$) to the original formant frequencies for each vowel. The results are the following: (1) perceived vowel height articulation shifts upward when the $F0$ shifts upward, while all formant frequencies remain the same: (2) this shift in vowel height is more distinct amid mid and low vowels than for high vowels; and (3) vowel height does not change when the $F0$ as well as all formant frequencies are shifted upward the same amount along the Bark scale. Further results, along with the hypothesis that a high $F0$ is regarded as the first formant in middle and low vowel perception, will be discussed.

WW12. Frication duration and amplitude rise time as cues to the voiceless fricative/affricate distinction. Margaret A. Walsh (Department of Psychology, University of Texas, Austin, TX 78712), Keith R. Kluender (Department of Psychology, University of Wisconsin, Madison, WI 53706), and Randy L. Diehl (Department of Psychology, University of Texas, Austin, TX 78712)

This paper describes the perceptual role of frication duration and amplitude rise time in signaling the fricative/affricate distinction in initial position. Sets of edited natural tokens of /ʃa/ and /tʃa/ were created in which fricative duration and rise time varied orthogonally. In one experiment, frication duration varied from 100 to 210 ms, and rise time was fixed at either 30 or 80 ms. Frication duration proved to be a robust cue for the fricative/affricate distinction, with longer durations yielding more /ʃ/ responses. Moreover, the longer value of rise time shifted the /ʃ/–/tʃ/ labeling boundary toward shorter frication durations; thus rise time had an enhancing effect on the perception of the fricative-duration cue. In a second experiment, rise time varied from 20 to 110 ms, while frication duration was fixed at either 140 or 160 ms. Although the fricative-duration parameter had a reliable effect on the percentage of fricative responses, variation in rise time alone had very little effect. These results are analogous to those of Walsh and Diehl [J. Acoust. Soc. Am. Suppl. 1 **82**, S80 (1987)], who found that transition duration plays a far more significant role than rise time in signaling the stop/glide distinction. [Work supported by NINCDS.]

WW13. Delayed pitch fall in Japanese: Perceptual experiment. Kazue Hata (Speech Technology Laboratory, 3888 State Street, Santa Barbara, CA 93105) and Yoko Hasegawa (Department of Linguistics, University of California, Berkeley, CA 94720)

Hasegawa and Hata [J. Acoust. Soc. Am. Suppl. 1 **83**, S29 (1988)] investigated the delayed pitch fall phenomenon in Tokyo dialect Japanese in production data and examined the relationships of perceived accent to the peak location and the steepness of the falling contour. Pitch fall, the only acoustic correlate of the accent in Japanese, sometimes occurred on the syllable following the accented syllable. A delayed pitch fall tends to be steeper the later it occurs. The present paper examines delayed pitch fall from a perceptual point of view. Three-syllable synthetic stimuli /ma ma ma/ were prepared with the $F0$ peak in different locations and different falling slopes. These stimuli were presented to native Japanese subjects in order to determine whether perception and production are correlated in this phenomenon, i.e., whether a change in the location of the peak and the degree of $F0$ fall in the second /ma/ causes the accent to be perceived on the first syllable. Implications for speech recognition will be discussed.

WW14. Some auditory–visual interactions are post-categorical. Richard E. Pastore, Jody K. Layer, Robert J. Logan, Stuart A. Tousman, and Hanson Hsu (Department of Psychology, SUNY Binghamton, Binghamton, NY 13901)

The McGurk effect recently has been variously conjectured to be based upon a precategorical interaction or integration of information from the auditory and visual representation of the linguistic event. The conjecture of precategorical interactions, which rejects the original notion of post-categorical interactions, may be in the form of a specialized module for speech or language, or may contribute to a logic-decision process, either of which leads to a relatively discrete perception of speech. While not disputing such precategorical interactions for language, the research to be described demonstrates post-categorical interaction of auditory and visual representations of phonemes. The stimuli are nonword CVC syllables. The set of auditory stimuli is edited from natural speech, while the set of visual stimuli includes the orthographic representations of the auditory stimuli syllables. Evidence for interactions in identification of initial phonemes is based upon both error rates and reaction times. [Research supported in part by an NSF grant to the first author.]

WW15. Can speech envelopes' modulation spectra be used to support segmental decisions? M. M. McCormick, R. J. Porter,[a] F. Seitz, and I. M. C. Watson (Oxford University Phonetics Laboratory, 41 Wellington Square, Oxford OX1 2JF, United Kingdom)

The auditory system senses the rate, the magnitude, and the phase of temporal modulations of sounds' spectra and amplitudes. Such "modulation sensations" have been proposed as a possible basis for perception of speech segments and their features [see R. J. Porter, in *Language Perception and Production*, edited by Allport *et al.* (Academic, London, 1987), Chap. 5]. This preliminary study extends a Russian attempt to analog-model speech segmentation decisions using speech-envelope modulation magnitude [Malinnikova *et al.*, Fiziol. Zh. SSSR im. I. M. Sechenova **66**, 139–145 (1980)]. In the current study, comparisons are made between human segmentation decisions and a multistage, computer-based, spectral analysis of the modulations of the amplitude envelopes within different speech spectral regions. [Research supported in part by Alvey Grant No. MMI092.] [a] Also at Kresge Hearing Research Laboratory, LSU Medical Center, New Orleans, LA 70112-2234 and Department of Psychology, University of New Orleans, New Orleans, LA 70112.

WW16. Cross-syllabic-position failures of adaptation are not due to acoustic–phonetic cancellation. Arthur Samuel (Department of Psychology, Box 11A Yale Station, New Haven, CT 06520)

Why does adaptation with a syllable-initial consonant fail to affect perception of the same consonant in syllable-final position, and vice-versa? One account of this well-replicated result invokes a cancellation

explanation [Pisoni and Tash, Percept. Psychophys. **18**, 401–408 (1975)]: With the place of articulation stimuli used, the pattern of formant transitions switches with syllabic position, allowing putative phonetic level effects to be opposed by putative acoustic level effects. Three experiments to be reported tested the cancellation hypothesis by preempting the possibility of acoustic countereffects. In experiment 1, the test syllables and adaptors were /r/–/l/ CVs and VCs that do not produce canceling formant patterns across syllabic position. In experiment 2, /b/–/d/ continua were used in a paired-contrast procedure, believed to be sensitive to phonetic, but not acoustic, identity. In experiment 3, cross-ear adaptation, also believed to tap phonetic rather than acoustic processes, was used. All three experiments refuted the cancellation hypothesis. Instead, it appears that the perceptual process treats syllable-initial consonants and syllable-final ones as inherently different. [Work supported by AFOSR.]

WW17. The perceptual distance between synthetic /s/ and /ʃ/ syllables in different vocalic contexts. Margaret F. Cheesman and Dianne J. Van Tasell (Department of Communication Disorders, Shevlin Hall, University of Minnesota, Minneapolis, MN 55455)

A nine-step, synthetic, /s/–/ʃ/ continuum was crossed with a five-step continuum of /i/ to /u/-like vowels to form 45 CV syllables. Thirteen listeners categorized the fricative portion of these syllables as /s/ or /ʃ/ in a 2AFC task. Identification judgments were strongly influenced by the vowel context; for example, more fricatives were identified as /s/ when they preceded /u/ than when they preceded /i/. The same group of listeners used a triadic comparison procedure [Levelt *et al.*, Br. J. Math. Stat. Psychol. **19**, 163–179 (1966)] to provide estimates of the perceptual distances between a subset of the fricatives in different vowel contexts. Perceptual distances were smaller for within-category pairs than they were for across-category pairs. Relations between the perceptual distance and the acoustic similarity of stimuli will be discussed. [Supported by NIH Grant No. NS12125, the Bryng Bryngelson Research Fund, and an SSHRC doctoral fellowship to MFC.]

WW18. Discrimination of level versus nonlevel pitch contours. George D. Allen (Department of Audiology and Speech Sciences, Purdue University, West Lafayette, IN 47907)

Phonetic transcription of intonation often requires a decision as to whether the pitch contour is rising, falling, or level. Although automatic procedures can measure voice fundamental frequency ($F0$) contours accurately, it is probably more appropriate to categorize them on the basis of what one's ears can discriminate. The fact that glottal periodicity includes jitter adds to the uncertainty of this discrimination. Pulse trains with level and rise–fall $F0$ contours were presented for discrimination by listeners in a 2I-FC paradigm. Within each pair, the $F0$ of the nonlevel began low, rose linearly for half the total duration, and then fell linearly to the end. Mean $F0$ was thus equal for the two items. Preliminary results using one duration (600 ms), three mean $F0$ levels (100, 125, and 150 Hz), three $\Delta F0$ values (1%, 2%, and 4%), and four jitter amounts (0%, 0.5%, 1%, and 2%) indicate that discrimination increases with increasing $\Delta F0$ and decreases with increasing jitter. Both trends were quite linear across the range of parameters employed in this preliminary study. The PEST procedures are now being used to map these relationships more precisely.

WW19. Identification of Polish words with non-neutralized word-final segments. Louisa M. Slowiaczek and Helena Szymanska (Department of Psychology, Loyola University of Chicago, 6525 North Sheridan Road, Chicago, IL 60626)

Research examining the phonetic characteristics of a number of neutralization rules has found that underlying contrasts that should be neutralized are phonetically preserved. In particular, earlier results [L. Slowiaczek and D. Dinnsen, J. Phonet. **13**, 325–341 (1985)] found evidence that the rule of word-final devoicing in Polish is not neutralized. The present investigation extended these production results by testing whether the acoustic measures identified in productions from the original study are functional in perception. Native Polish and English listeners identified Polish monosyllabic words using a two-alternative forced-choice procedure. Results for the Polish subjects indicated a bias to choose the voiceless alternative and suggested that Polish listeners are unable to perceive differences in the minimal pairs examined in the production study. [Work supported by a small grant from Loyola University of Chicago.]

WW20. Auditory memory in phonetic and nonphonetic judgments of vowels. Sumi Shigeno (Liberal Arts and Sciences, Kitasato University, 1-15-1 Kitasato, Sagamihara, 228 Japan)

The present study compares the magnitudes of context effects in the perception of phonetic and nonphonetic features of stationary vowels. The stimuli were synthetic vowels on the [u]–[a] continuum generated by varying stimultaneously $F1$ (442–586 Hz) and $F0$ (118–154 Hz). Thus vowel [a] had a higher pitch than [u]. Category boundaries were determined for the following three conditions: (1) isolated vowels presented in random sequence, (2) vowels with a preceding context stimulus [u], and (3) with white noise inserted between the context stimulus and the target stimulus. When the ISI between the context and the target was 2.0 s, the context effect was contrasted in the case of phonetic judgment, and its magnitude was reduced by the insertion of white noise. It was assimilated in the nonphonetic judgment, and its magnitude was increased by the insertion of noise. When the ISI was 0.5 s, the magnitude of the assimilation in the nonphonetic judgment was not increased by the insertion of noise. These results suggest that the reduction of contrast was not due to a decrement in auditory memory, which can be considered as a fast-decaying component.

WW21. Effects of syllable duration on the perception of Mandarin tones: A cross-language study. Deborah L. Blicher, Randy L. Diehl, and Leslie B. Cohen (Department of Psychology, University of Texas, Austin, TX 78712)

Mandarin syllables carrying a high-rising $F0$ contour (tone 2) tend to be produced shorter than those carrying a low-falling–rising contour (tone 3). It is suggested that, even for syllables with the same physical duration, the more complicated $F0$ structure of tone 3 makes it appear longer than tone 2. Talkers may therefore produce an actual length difference to enhance the apparent length difference. This hypothesis was tested by comparing perceptual judgments by native Mandarin and native English speakers on various series of Mandarin tones. A short-stimulus (350 ms) series and a long-stimulus (450 ms) series were synthesized for each of the syllables /bi/, /ba/, and /bu/. Each series was generated by incrementally interpolating the $F0$ contour between a tone 2 and a tone 3 exemplar, both of which were Mandarin morphemes or words. Subjects (both Mandarin- and English-speaking) were first trained with feedback to assign the short and long series-endpoint stimuli to two categories based on $F0$ contour alone. Next, subjects identified the entire stimulus series (both long and short) on the basis of the two training categories. For both Mandarin- and English-speaking subjects, a longer syllable duration shifted the labeling boundary reliably toward fewer tone 2 (i.e., more tone 3) responses. The parallel boundary shifts suggest that length variation enhances the perceptual distinction between tones 2 and 3, probably by reinforcing what is already a difference in apparent length. [Work supported by NINCDS.]

WW22. Phoneme processing in the perception of spoken Japanese words. Shigeaki Amano (NTT Basic Research Laboratories, 3-9-11 Midori-cho, Musashino, 180 Japan)

Cohort theory, a speech perception model based on serial phoneme processing, is studied in respect to the lexical access of Japanese words. Six subjects made a lexical decision by pushing a key as soon as possible after hearing real Japanese words or nonwords that were 2–5 mora long and had various nonword discrimination points. A nonword discrimination point was defined as a point where no other real word candidates could be found in a cohort by serial phoneme processing. The results show that the reaction times measured from the nonword discrimination point are not constant. The nearer the nonword discrimination point is to the end of the word, the shorter the reaction times are. This indicates the existence of parallel phoneme processing. Moreover, the reaction times for short words are longer than for long words when they are measured from the end of the word. This shows that short words are buffered somewhere in the word perception process until the lexical decision is made. However, there is also evidence of serial processing. The reaction times from word onset gradually increase along with time to the nonword discrimination point. These results suggest that phoneme processing is quasiserial. Parallel or buffered processing should be incorporated into cohort theory along with serial processing.

WW23. Naturalness and intelligibility of amplitude modulated time-varying sinusoidal speech. Thomas D. Carrell (Department of Sciences and Disorders, Northwestern University, Evanston, IL 60208)

Synthetic speech has been used for decades to test theories regarding human speech perception. Typically, the speech has been constructed to sound as natural as possible given the constraints of the experiment. An alternative strategy has been to examine the perception of intentionally impoverished stimuli such as time-varying sinusoid (TVS) replicas of speech [Remez *et al.*, Science **212**, 947–950 (1981)]. The TVS signals consist of three tones whose frequencies mimic the formant center frequencies of a natural sentence. They exhibit few of the acoustic properties of natural speech. It has been demonstrated that TVS signals sound extremely unnatural although they are surprisingly intelligible. The goal of the present experiment was to determine some of the general acoustic characteristics of signals that are important for speech perception. This was accomplished by examining the perceptual consequences of adding simple temporal and spectral information to TVS sentences. The TVS signals were amplitude modulated at 100 Hz in order to give them more speechlike acoustic characteristics without giving them fundamental frequencies or harmonic structures. The modulation greatly improved the phonetic intelligibility of the acoustically sparse TVS signal. The modulated signal was also significantly more natural sounding to listeners than the unmodulated TVS signal. Performing this operation on natural speech, however, caused a decrement in intelligibility.

WW24. The role of attention in speech perception. Bertram Scharf, Huanping Dai, and Joanne L. Miller (Psychology Department, Northeastern University, Boston, MA 02115)

This study investigated the role of auditory attention during speech perception. The syllables /da/ and /ga/ were synthesized so that they differed in initial burst and third formant transition; the critical distinguishing information was in the vicinity of 2.5 kHz. Discrimination was first measured with a 1I, 2AFC procedure under six masking conditions. Performance was near chance (55% correct) when the masker was centered at 2.5 kHz, but increased as the masker moved away from this critical frequency region, reaching 100% with the masker at 1 kHz. Next examined was whether listeners' attention as they performed the task was focused specifically on the 2.5-kHz region, or spread across all frequency regions. In one condition, subjects were asked to discriminate the syllables when a weak 90-ms, 1-kHz tone was added to /da/ and, in the other condition, when a weak 90-ms, 2.5-kHz tone was added to /da/; subjects were not informed that the tones had been added. In both conditions, the masker was centered at 2.5 kHz. Performance was at 58% when the 1-kHz tone was added, but at 75% when the 2.5-kHz tone was added. In two control conditions, it was found that, when subjects were informed that the tones had been added, so that they could focus their attention on the relevant frequency regions, performance increased substantially (to 90% correct) for the 1-kHz condition, but only slightly (to 81% correct) for the 2.5-kHz condition. These results suggest that, when attempting to discriminate syllables, listeners focus their attention on the specific frequency region critical to the distinction. [Work supported by NIH.]

WW25. The neural coding of relational invariance in speech: Human language analogs to the barn owl. Harvey M. Sussman (Department of Linguistics and Speech Communication, University of Texas, Austin, TX 78712)

This paper presents a brain-based perspective on the noninvariance problem in speech research, namely, that physically different auditory stimuli come to elicit an invariant perception of a phoneme category. Extrapolating from the sound localization system in the barn owl, two speculative human models are offered that account for the emergence of relational invariance for (1) bilabial and alveolar stop consonants across vowel contexts, and (2) vowel identity across speakers with different sized vocal tracts. The barn owl's extraction of interaural time differences (ITDs) to signal azimuth is based on disambiguating interaural phase information coded by frequency specific neurons. Only by spanning across a broad frequency spectrum can a neuronal functional array signal an unambiguous ITD to higher centers. A similar principle is invoked to model both stop consonant place invariance and vowel normalization in human speech processing. Formant manipulation metrics, capable of distinguishing contrastive phonemic categories, are described to illustrate the operational features of the modeling scheme.

WW26. Acoustic correlates of intervocalic stop confusions. H. Timothy Bunnell (CASS, Gallaudet University, Washington, DC 20002) and James G. Martin (Department of Psychology, University of Maryland, College Park, MD 20732)

As part of the study of signal processing algorithms that automatically enhance speech intelligibility, the acoustic factors underlying place confusions in intervocalic stop consonants have been examined. The speech database in these studies consists of nine repetitions of each of eighteen nonsense utterance frames from five talkers. The utterance frames were of two forms: "say a CVwuh" and "say a wVCuh" with C = {/b/,/d/,/g/} and V = {/i/,/a/,/u/}. In producing the repetitions of each frame, talkers were under instructions to vary their speaking style from enunciating very clearly to speaking in a casual, relaxed fashion. Confusion matrices and percentage correct consonant identification have been obtained for these utterances from both normal-hearing listeners (using broadband noise as a masker) and from hearing-impaired listeners. Acoustic measures related to place of articulation cues and general spectral/temporal contrast have been obtained from each utterance. The relations between these acoustic measures and consonant confusability will be described and implications for algorithms intended to enhance speech intelligibility will be discussed. [Work supported by NINCDS.]

WW27. Some parameters affecting the perception of voicing for fricatives. Laura B. Glicksman, Kenneth N. Stevens (Research Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge, MA 02139), and Sheila E. Blumstein (Department of Cognitive and Linguistic Sciences, Brown University, Providence, RI 02912)

Measurements of voicing have shown that low-frequency periodicity can usually be found at VC and CV boundaries, specifically in the initial 30 ms or the final 20 ms of a phonetically "voiced" fricative. The purpose of this study is to determine the perceptual importance of voicing at VC and CV boundaries for fricatives. The VCV tokens with alveolar fricatives were synthesized with various fricative durations and various patterns of voicing within the consonant. Listeners identified the consonant as either [s] or [z]. Short fricatives are heard as [z] but, for durations longer than 70 ms, voicing must occur at either the VC or CV boundary. Voicing placed in the middle of the fricative did not elicit [z] responses. When voicing occurred at the boundaries, perception was influenced by fricative duration, and by amplitude and duration of voicing. Possible theoretical interpretations of the data will be discussed. [Work supported by grants from NINCDS to Brown University and to MIT.]

WW28. Context-independent dynamic information for vowel identification. Winifred Strange and James J. Jenkins (Departments of Communication Sciences and Disorders and Psychology, University of South Florida, Tampa, FL 33620)

Previous studies have demonstrated that listeners can identify vowels spoken rapidly in stop consonant–vowel–stop consonant syllables even when vocalic nuclei of the acoustic signals are attenuated to silence, leaving only initial and final transitions. In this study, mixed consonant "silent-center" (MC) conditions were generated in which the initial and final transitions of syllables containing the same vowel but different stop consonants were interchanged, thus altering context-dependent formant trajectories. In one condition, silent intervals were appropriate for the original vowel and the final consonant; in a neutral duration MC condition, silent intervals were adjusted so that intrinsic vowel duration cues were eliminated. Identification errors in the MC condition (8%) were not significantly higher than for original silent-center syllables; the neutral duration MC condition produced somewhat more errors (15%), but identification was more accurate than for initial transitions (35% errors) or final transitions (52% errors) presented alone. These results suggest that some perceptually relevant aspects of dynamic information are context independent. Representations of formant trajectories presented in $F1$–$F0$/$F3$–$F2$ "space" demonstrate context-dependent and -independent aspects of dynamic information. [Work supported by NINCDS.]

WW29. Effect of vowel duration on the perception of syllable-initial /ʃ/ and /tʃ/. Keith R. Kluender (Department of Psychology, University of Wisconsin, Madison, WI 53706) and Margaret A. Walsh (Department of Psychology, University of Texas, Austin, TX 78712)

The duration of an adjacent vowel has been demonstrated to affect the judgment of consonant duration and, hence, phonetic identity. For example, syllable-initial stops can be distinguished from glides on the basis of transition duration and, when the following vowel is relatively long, longer transitions are required in order for a glide to be perceived. Apparently, the vowel provides a source of durational contrast, whereby a longer vowel makes the adjacent consonant seem shorter. This effect has been demonstrated for both speech and nonspeech signals [L. Diehl and M. A. Walsh, J. Acoust. Soc. Am. Suppl. 1 **80**, S125 (1986)], and is presumably grounded in a general auditory mechanism. In this study, the effect of vowel duration on perception of syllable-initial consonants is also demonstrated for fricatives and affricates. Frication duration is an important cue to this distinction, with longer durations yielding more /ʃ/ responses. Two series of edited natural tokens of /ʃa/ and /tʃa/ were created in which frication duration varied from 100 to 210 ms. The duration of the following vowel was 204 ms for one series, and 358 ms for the other. With the longer vowel context, significantly longer frication was required to yield a /ʃ/ percept. Consistent with earlier findings for other consonantal distinctions, durational contrast occurred between the vowel and fricative. [Work supported by NINCDS.]

WW30. Perception of fluent and nonfluent speech in spontaneous utterances and spoken text. Peter Homel and Laura Brighenti (Institute for Speech and Language Sciences, New York University, New York, NY 10003)

Modeled after Remez *et al.* [J. Acoust. Soc. Am. **77**, 538 (1985)], this study measured the perceptual differences between spontaneous and read versions of sentences, where read versions were made to sound "spontaneous." Subjects were required to distinguish between sentences selected from a recorded interview and versions of those sentences from a recording of the speaker reading a highly faithful transcript of the original interview (including emphasis, pauses, and dysfluencies). Sentences selected for the perception experiment were of two types: intact, well-formed fluent sentences and nonfluent sentences containing some minor dysfluency but which are otherwise well formed. Results showed that subjects performed at a level better than chance but lower than that found in Remez *et al.* Subjects also tended to show better performance for nonfluent sentences, in agreement with subjects' reports that timing and perceived anticipation of dysfluencies seemed to distinguish spontaneous and read versions. These results replicate and extend Remez *et al.*, and point out the possible role that dysfluencies play in the perceived spontaneity of speech.

WW31. An audiovisual investigation of the loudness/effort effect for speech and nonspeech events. Lawrence D. Rosenblum (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511)

There is some evidence that loudness judgments of speech are more closely related to the degree of vocal effort induced in speech production than to the speech signal's surface acoustic properties, such as intensity [P. Ladefoged and N. P. McKinney, J. Acoust. Soc. Am. **35**, 454–460 (1963)]. Other researchers have claimed that speech loudness can be rationalized by considering simply the acoustic complexity of the signal [R. D. Glave and A. C. M. Reitveld, J. Acoust. Soc. Am. **58**, 875–879 (1975)]. Since vocal effort can be specified optically as well as acoustically, a study to test the effort–loudness hypothesis was conducted that used conflicting audiovisual presentations of a speaker producing CVs with different efforts. The prediction was made that if loudness judgments are based on effort perception rather than on simple acoustic parameters, then judgments should be affected by visual as well as auditory information. It is shown that loudness judgments are significantly affected by visual information even when subjects (1) are instructed to base their judgments on only what they hear and (2) cannot detect a discrepancy between the audio and visual components. Moreover, the same results are shown for a nonspeech "clapping" event attesting to the generality of the loudness–effort effect previously thought to be special to speech.

THURSDAY PM

WW32. Effects of directed attention on the reliability of dichotic listening. Ingrid M. Blood and Gordon W. Blood (Department of Communication Disorders, The Pennsylvania State University, University Park, PA 16802)

This investigation examined the performance of adult subjects on perception of dichotic CVs. Twenty subjects were tested over five separate sessions to determine the reliability of responses. Instructions utilized either free recall or directed listening to the right or left ear. The methods of analysis used were d' (% RC − % LC), laterality equation (R − L/R + L × 100), percent of error (LE errors/total errors), and phi coefficient ($\mathrm{R} - \mathrm{L}\sqrt{\mathrm{R} + \mathrm{L}[2T - (\mathrm{R} + \mathrm{L})]}$). Results indicated that the right ear advantage remained consistent for the right, left, and no direct condition. Differences in dichotic performance were found over the five trials.

WW33. Using reaction times to explore the mechanisms of selective adaptation. Donna Kat and Arthur G. Samuel (Department of Psychology, Yale University, Box 11A Yale Station, New Haven, CT 06520)

Previous research has demonstrated that the identification shifts induced by selective adaptation are accompanied by reliable reaction time changes. For example, if a /ba/–/da/ test series is used, reaction times to tokens at the /b/ end of the series will be higher after /ba/ adaptation than after /da/ adaptation; at the /da/ end, the reverse is true [A. G. Samuel, Cog. Psychol. **18**, 452–499 (1986)]. The present study extends these results in three ways. First, a neutral adaptation condition (/a/) is used to provide an appropriate baseline for measuring *absolute* reaction time changes. Second, combination adaptors (/ba/–/da/) are used to look for reaction time changes in the absence of labeling shifts. Third, cross-series reaction time changes (/pa/,/ta/, and /pa/–/ta/ on a /ba/–/da/ series) are examined. The pattern of reaction time changes will be used to address the level of processing affected by adaptation, and to clarify the mechanism of the effects (contrast versus fatigue).

THURSDAY AFTERNOON, 17 NOVEMBER 1988

IAO NEEDLE/AKAKA FALLS ROOM, 1:30 P.M.

Meeting of Accredited Standards Committee S1 on Acoustics

to be held jointly with

Technical Advisory Group Meeting for ISO/TC 43 Acoustics, IEC/TC 29 Electroacoustics, and IEC/TC 87 Ultrasonics

D. L. Johnson, Chairman S1

Larson-Davis Laboratories, 280 South Main, Pleasant Grove, Utah 84062

Standards Committee S1 on Acoustics. Working Group chairs will report on their progress in the preparation of standards, methods of measurement and testing, and terminology in physical acoustics, electroacoustics, sonics, ultrasonics, and underwater sound. Work in progress includes measurement of noise sources, noise dosimeters, integrating sound-level meters, and revision and extension of sound-level meter specifications. Open discussion of committee reports is encouraged.

The international activities in ISO/TC 43 Acoustics, IEC/TC 29 Electroacoustics, and IEC/TC 87 Ultrasonics will be discussed. The chairs of the respective U.S. Technical Advisory Groups for ISO/TC 43 (H. E. von Gierke), IEC/TC 29 (V. Nedzelnitsky), and IEC/TC 87 (P. D. Edmonds) will report on current activities of these Technical Committees.

Reports will be given on the meeting of ISO/TC 43 and IEC/TC 29, taking place in Toronto, Canada in October 1988, and on that of IEC/TC 87, meeting in Philadelphia, PA from 24–28 October 1988.

Session XX. Musical Acoustics IV: Musical Instruments East and West II: Wind and String Instruments

Tohru Idogawa, Cochairman
Institute of Applied Physics
University of Tsukuba
Tsukuba, 305 Japan

William J. Strong, Cochairman
Department of Physics
Brigham Young University
Provo, Utah 84602

Chairman's Introduction—2:00

Invited Papers

2:05

XX1. An experimental study on the reed vibrations of the woodwind instruments. Tohru Idogawa, Masakazu Iwaki, Toshikatsu Naoi, and Michiko Shimizu (Institute of Applied Physics, University of Tsukuba, Tsukuba, 305 Japan)

Reed vibrations have been observed for the lowest notes of a bassoon, an oboe, a clarinet, and a soprano saxophone artificially blown. The blowing pressure in the artificial mouth used has been increased from 1.5 kPa to about 10 kPa (20 kPa for the oboe) and decreased. The relative position and contact pressure between the lips and reeds have been fixed during a cycle of changes in the blowing pressure. Conclusions: (1) Many different modes of very stable periodic vibrations are found even under the same blowing pressure. (2) A slow continuous change in blowing pressure causes a sudden transition from one mode to the next without a change in the lip–reed relative position; this in turn gives rise to some change in the blowing pressure. (3) Which mode is excited under a given blowing pressure depends on the sequence of the preceding modes. (4) In the bassoon and oboe, not all the transitions are reversible; with blowing pressure being increased and decreased, the sequence of modes encountered exhibits hysteresis. (5) Wolf-tone-like waveforms have been found in all instruments used.

2:30

XX2. On the modeling of self-oscillation in brass instruments. Shigeru Yoshikawa (5th Research Center, Technical R&D Institute, Japan Defense Agency, Nagase, Yokosuka, 239 Japan)

It is recognized that the lips of brass players vibrate two-dimensionally, i.e., in parallel and perpendicularly to the direction of the air flow as the excitatory source. This two dimensionality gives two different models of lip motion in brass instruments: (1) the parallel model as a "reed striking outwards," which is the reverse to the model of a woodwind reed as a "reed striking inwards" [N. H. Fletcher, Acustica **43**, 63–72 (1979)], and (2) the perpendicular model as a "one-mass vocal cord striking laterally" [J. Saneyoshi, H. Teramura, and S. Yoshikawa, Acustica **62**, 194–210 (1987)]. The phase difference between the lip displacement and the acoustic pressure acting on the lips, or the frequency difference between the antiresonance of the horn itself and the sound excited, can determine which model is actually dominant. This paper describes some attempts to find these quantities on a cylindrical pipe blown by lips without a mouthpiece. The discussion of the experimental results and the modeling of the lip vibration is presented after briefly explaining the differences between the regeneration theories based on the above models.

2:55

XX3. Computational study of brass instrument configurations. J. Duane Dudley and William J. Strong (Department of Physics and Astronomy, Brigham Young University, Provo, UT 84602)

Input impedance curves are computed for several bores related to brass wind instrument shapes, such as a cylinder, cone, French horn bell, "catenoidal bell," some "bell + cylinder" combinations, and a complete French horn. The impulse response curve for each is obtained by performing a Fourier transform of the complex input impedance. These will be shown and discussed with respect to physical interpretations and interesting insights. A sinusoidal input volume velocity is then introduced, and the steady-state (sinusoidal) pressure response at the inlet is determined by convolving the velocity input with the impulse response. This is

done for various frequencies, and the resulting pressure amplitudes are shown to be consistent with those of the impedance curve. An input volume velocity composed of several harmonically related sinusoidal components, approximating the output from a player's lips, is then introduced. The resulting pressure response is computed at the inlet of a French horn, whose resonant frequencies are quite harmonic. The same computations are then made for a relatively inharmonic system, and the results are compared to those for the French horn.

3:20

XX4. Wind instrument bores in the time domain. R. Dean Ayers (Department of Physics–Astronomy, California State University, Long Beach, CA 90840)

Recent interest in modeling nonlinear feedback control of the driver has led several researchers to examine Green's functions and reflection impulses for wind instrument bores. These are usually obtained as inverse Fourier transforms of experimental or computational results in the frequency domain. At CSULB, the focus instead has been on developing such waveforms stepwise, using multiple convolutions. This was motivated by the desire to obtain a better intuitive understanding than that provided by just the transmission line model in the frequency domain. Central to this work are such basic concepts as causality and round trip delay time. The intuitive power of this theoretical approach correlates nicely with the cleanness of experimental results obtained directly in the time domain, which allow us to isolate the effects of individual bore elements. The experimental technique employs Wiener filtering to generate a brief acoustic pulse from a piezoelectric element. Resultant waveforms are further processed by a Scientific Atlanta SD-380 spectrum analyzer, with data sent backward through an IBM-PC to average out low-frequency noise. [Work supported by the CSULB Foundation.]

3:45–4:00
Break

4:00

XX5. Nonlinear theory of musical wind instruments. Neville H. Fletcher (CSIRO and Research School of Physical Sciences, Australian National University, Canberra, ACT 2601, Australia)

While many important features of the design and playing of musical wind instruments can be appreciated from a linear theory of their acoustical operation, an understanding of tone quality, loudness, and transient behavior depends almost entirely upon the essential nonlinearity of the sound generation mechanism and its interaction with resonances of the horn. This has been appreciated in a general way since the time of Bouasse, early this century, while Benade has led the renaissance of interest in the topic over the past 25 years. Until recently, available theoretical methods allowed the calculation of behavior only for rather soft playing conditions and consequently for limited nonlinearity. The development of time-domain and mixed-domain "frequency-balance" techniques during the past few years, however, now allows all regimes of playing to be investigated. This paper reviews available theoretical methods and their application to typical instruments.

4:25

XX6. Kinematical and dynamical factors for deviation from Helmholtz motion. Gabriel Weinreich (Randall Laboratory of Physics, University of Michigan, Ann Arbor, MI 48109)

In the ideal Helmholtz motion, the string velocity at the bow alternates between two constant values, the ratio of whose magnitudes is equal to the ratio of lengths of the two segments into which the bow divides the string. Deviations from this ideal can be of two types: (a) motions for which the string behavior at the bow maintains its "two-velocity" nature but with a different ratio; and (b) motions that deviate from the "two-velocity" pattern. Kinematically, two-velocity motions of arbitrary ratio are possible if the bowing point is not a rational subdivision of the string; otherwise, if it is defined as a fraction m:n of two integers in its lowest terms, the shortest possible duty cycle (that is, fraction of period for which "slipping" occurs) is $1/n$. Small ratios may be favored by stability considerations. Deviations from the two-velocity pattern must occur when dissipation is present, and involve the response of the string to near-resonant periodic forces. Examples observed with the "digital bow" will be shown. [Work supported by NSF.]

4:50

XX7. New solutions of the wave equation for a bowed string. Takashi Kojima (Faculty of Engineering, Shizuoka University, 3-5-1 Jouhoku, Hamamatsu, 432 Japan)

In 1860, H. v. Helmholtz concluded, by observing the movement of each particular point of a bowed string with his vibration microscope, that the highest point of the bowed string travels along upper and lower

parabolic arcs. Since then, his conclusion has been accepted and little attention has been paid to the shape of the envelope of the bowed string. The experimental work of Kondo *et al.*, which focused on this subject [6th I.C.A., N-2-4 (1968)], stimulated the present work. This work re-examines the general solution of the wave equation by giving some initial conditions and then, in addition to the normal Helmholtz solution, obtaining two new solutions, whose shapes are of an elliptic arc and of hyperbolic arc, respectively. These three solutions, parabolic, elliptic, and hyperbolic, coincide with each other when a specific parameter approaches infinity. One experimental result found in the literature seems to represent an elliptic solution.

5:15

XX8. Measurement of the bowing parameters in violin playing. Anders Askenfelt (Department of Speech Communication and Music Acoustics, Royal Institute of Technology, Box 70014, S-100 44 Stockholm, Sweden)

A method is presented that allows the measurement of the bowing parameters in violin playing, without interfering with normal playing conditions. The measured parameters include the force between bow and string ("bow pressure"), the position of a contact point between bow and string along the bow ("bow position"), and the distance between the contact point and the bridge ("bow–bridge distance"). Typical registrations for a sample of bowing patterns will be presented, and the violinist's use of the bowing parameters in controlling the dynamic level will be discussed.

THURSDAY AFTERNOON, 17 NOVEMBER 1988 — WAIANAE ROOM, 2:00 TO 5:00 P.M.

Session YY. Physical Acoustics VI and Bioresponse to Vibration IV: Mechanisms of Biological Response to Ultrasound and Vibration

Wesley L. Nyborg, Cochairman
Physics Department
University of Vermont
Burlington, Vermont 05405

Shigeo Ohtsuki, Cochairman
Research Laboratory of Precision Machinery and Electronics
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4259 Nagatsuta, Midori-ku
Yokohama, 227 Japan

Invited Papers

2:00

YY1. Cell manipulation by radiation forces. W. Terence Coakley, David W. Bardsley, and Martin A. Grundy (Department of Microbiology, University College, Cardiff CF2 1TA, United Kingdom)

Particles suspended in acoustic fields experience nonzero time averaged radiation forces that can cause particle movement, interparticle attraction or repulsion, and can exert a torque on suspended particles. In a standing wave, such field forces result in migration of the particles to positions separated by half an acoustic wavelength. The forces depend on some or all of the following properties: particle size and shape, sound frequency, the square of the sound-pressure amplitude, and the differences between the density and compressibility of the particles and those of the suspending phase. Threshold pressure for the vertical migration of particles to half-wavelength separations is derived. It is shown that appropriate choice of sound-pressure amplitude should lead either to the separation of particles of different densities and compressibilities or to the concentration of mixed samples essentially at the same location. Applications of radiation forces to bring cells together in suspension for cell electrofusion procedures and the use of ultrasound to support cells away from solid surfaces are described. Examples of cell concentration, cell sedimentation, cell alignment, and cell–cell interaction are also presented.

2:20

YY2. Advances in microparticle characterization using high-frequency (30 MHz) acoustic scattering. X. Chen, R. A. Roy, and R. E. Apfel (Department of Mechanical Engineering, Yale University, P.O. Box 2159, New Haven, CT 06520)

There are a variety of characterization techniques for individual and ensembles of particles. Of the former category, however, there are none that characterize microparticles (in the micron range) by the mechanical

descriptors: density and compressibility. An apparatus developed in the Yale Acoustics Laboratory accomplishes this objective, thereby allowing one to determine the distribution of a given property for a population of particles individually convected past the focal zone of confocally positioned acoustic transmitter and receivers. In earlier work, two- and three-transducer systems operating at 30 MHz have produced data sets that have allowed for the prediction of density and compressibility of particles (e.g., red blood cells) given *a priori* information about the size of these particles. More recently, the two-transducer acoustic apparatus has been married to a modified electrozone sensing cell (such as found in Coulter counters) that provides size information for each cell. Particles pass first through the electrozone unit and then through the focal zone of the acoustic unit. The result is that now the volume, compressibility, and density can be determined for each particle. Initial results and limitations of the apparatus will be presented. [Work supported by Grant R01-GM30419 of the U.S. National Institutes of Health.]

2:40

YY3. A flow-cytometric study on cultured cells after exposure to pulsed ultrasound. Kazuo Maeda and Junzo Kigawa (Department of Obstetrics and Gynecology, School of Medicine, Tottori University, Yonago, 683 Japan)

HeLa cells were cultured inside a flask that allowed ultrasonic penetration, and were exposed to 3, 5, or 10 μs wide experimental ultrasound, the intensity of which was measured by microbalance, though it will be further confirmed by the hydrophone technique. The exposure was made in a water bath for 15 min, and then the cells were analyzed with a Cytofluorograf 30H flow cytometer. The cell DNA distribution showed a change that was comparable to the effect of gamma-ray exposure, or a cytostatic agent after exposure to 5- or 10-μs SPTA 2.35 W/cm^2 ultrasound, but no change was observed after the exposure to 3-μs pulsed ultrasound with the above-mentioned intensity, or after exposure to 10-μs 0.2 W/cm^2 pulsed ultrasound. The results were similar to a study on the suppression of the cell growth curve after exposure to experimental ultrasound on JTC-3 cultured cells.

Contributed Papers

3:00

YY4. Acoustic streaming generated by surface oscillations. Charles Thompson and Vineet Mehta (Department of Electrical Engineering, Laboratory of Advanced Computations, University of Lowell, 1 University Avenue, Lowell, MA 01854)

An analysis of streaming in a oscillatory flow over a two-dimensional temporally and spatially varying boundary will be presented. This phenomenon plays an important role in transport of passive contaminants. Longuet–Higgins have shown the role played by Reynolds stress in producing drift in a viscous fluid. The modifications in streaming due to interplay between surface oscillations and the flow field affect the mass transport velocity. A primary objective in modeling the mammalian auditory system is an analysis of the fluid motion in the cochlea. The deflections of the basilar membrane, which produce the sensation of hearing, are coupled to the streaming flow. A theoretical model describing the fluid motion will be given. It will be shown that the dynamic behavior of the flow field is characterized by the parameters ϵ, a typical slope; R, the oscillatory Reynolds number; S, the Strouhal number; and α, the amplitude of the temporal variation. An analytic solution will be sought in terms of an expansion in ϵ. [Work supported by Analog Devices Professorship.]

3:12

YY5. Shock wave production in medical ultrasound—A new computational approach. P. T. Christopher and K. J. Parker (Rochester Center for Biomedical Ultrasound of Rochester, Department of Electrical Engineering, University of Rochester, Rochester, NY 14627)

At high intensities, the presence of nonlinear propagation effects can influence the performance of medical ultrasound devices. In general, models that can fully account for the combined effects of tissue nonlinearities, tissue attenuation, and beam diffraction effects are quite complex, require extensive computational time, and provide little physical insight as to the interaction of these processes. A novel, computationally efficient model has been developed that accounts for the effects of nonlinearities, attenuation, and diffraction. The model propagates a circularly symmetric beampattern forward in small axial increments. Over the increment, the fundamental and harmonics are diffracted and attenuated linearly using angular spectrum decomposition techniques. Then, the complex amplitudes of the fundamental and harmonics are altered according to theory of plane-wave propagation in a medium with nonlinear behavior. This hybrid, incremental algorithm is shown to reproduce theoretical Gaussian and focused piston beampatterns described by others, and also matches experimental data from focused ultrasound beampatterns measured in the present study. The new hybrid algorithm is computationally efficient and provides some insight into the relative importance of nonlinearities, attenuation, and focusing in medical imaging and lithotripsy conditions. [Work supported by NIH.]

3:24

YY6. Dynamic characterization of a radio ear bone vibrator with laser vibrometry. Martin J. Pechersky, A. D. Stuart, and M. R. Bufalini (The Applied Research Laboratory, The Pennsylvania State University, P.O. Box 30, State College, PA 16804)

The vibrational characteristics of a model B-71 bone vibrator were determined using a laser vibrometer. The bone vibrator is normally used for auditory evoked potential testing. The purpose of these measurements was to characterize the performance of this device between a frequency range of 500–5000 Hz. The dynamic characteristics of the bone vibrator were measured with the device mounted on a Beltone model M5B artificial mastoid. Frequency response functions were determined for the bone vibrator/mastoid system at several locations. Subsequently, transfer func-

tions between the driving voltage and the seismic mass of the bone vibrator, the case of the bone vibrator, and the artificial mastoid were determined, as well as the transfer function between the seismic mass and the mastoid. The vibrations of the bone vibrator were obtained with a high degree of spatial resolution by focusing the vibrometer laser beam on the location of interest to obtain the instantaneous velocity at that point. The mastoid response was measured with a conventional accelerometer. Besides the resonance associated with the seismic mass, two structural modes of the bone vibrator case were identified in the frequency range of interest and the mode shapes of the case vibrations were investigated.

3:36

YY7. Measurements of the response of gas-filled micropores to ultrasound by using PVDF transducers. Junru Wu and Wesley L. Nyborg (Department of Physics, University of Vermont, Burlington, VT 05405)

Techniques that use hydrophobic polycarbonate thin sheets containing randomly spaced, fairly uniform small holes, immersed in water to trap air bubbles, have been found useful in biophysical experiments. Using such sheets, the scattering from a stable array of small air-filled holes exposed to ultrasound has been investigated [D. L. Miller, J. Acoust. Soc. Am. **71**, 471–476 (1982)]. A set of piezoelectric crystal transducers of high-quality factor were used during the earlier study, which allowed measurements to be made at a discrete set of frequencies. In the present work, utilization of PVDF transducers, which have flat frequency response in the range of interest, makes it possible to study the continuous frequency response of the trapped bubbles. Results of the new experiments will be presented and discussed. [Work supported by institutional funds of University of Vermont.]

3:48

YY8. Effects of air gun energy releases on the northern anchovy. C. F. Greenlaw, D. V. Holliday (Tracor Applied Sciences, San Diego, CA 92123), R. E. Pieper (University of Southern California, Institute for Marine and Coastal Studies, Terminal Island, CA 90731), and M. E. Clark (University of Miami, Rosensteil School of Marine and Atmospheric Sciences, 4600 Rickenbacker Causeway, Miami, FL 33149)

Eggs, larvae of several ages, and adult specimens of the northern anchovy were exposed to acoustic energy from air gun seismic sources. Unlike earlier studies that exposed specimens to one or a few shots from the source, these exposures used a series of air gun shots using devices of various chamber sizes, simulating realistic exposures that might be experienced by wild fish in the vicinity of a seismic survey vessel. Exposure protocols ranged from the equivalent of single, distant passes of a typical seismic array to multiple passes at very short range. Air gun charging pressures were varied to simulate changing ranges. Histological examination of exposed specimens revealed no evidence of gross morphological damage caused by exposure. Comparison of survival with control groups showed subtle effects in the younger (2–4 days) larvae. Exposure of adults resulted in some damage to swimbladders, particularly for fish exposed at the surface where water particle motion effects are pronounced, but no significant effects on otoliths were found. Extrapolation of the survival and histology data suggests that noticeable impacts on eggs and larvae of this fish would result only from multiple, close exposures to seismic arrays. [Work supported by API and the California Egg and Larvae Committee.]

4:00–5:00
Discussion

THURSDAY AFTERNOON, 17 NOVEMBER 1988 KOHALA/KONA ROOM, 2:00 TO 3:30 P.M.

Session ZZ. Education in Acoustics I: Graduate Programs for Education in Acoustics (Poster Session)

David K. Holger, Cochairman
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Kobe, 657 Japan

Contributed Papers

Posters will be displayed from 2:00 to 3:30 p.m. Authors will be at their posters during the entire session.

ZZ1. The graduate program in architectural acoustics at the University of Florida. Bertram Y. Kinzey, Jr. and Gary W. Siebein (Department of Architecture, 231 ARCH, University of Florida, Gainesville, FL 32611)

The graduate program in architectural acoustics at the University of Florida offers a course of study that leads to Master of Architecture and Ph.D. degrees. The focus of the program is to address the theories and practice of acoustics in architectural settings. Core courses are offered in basic acoustical principles and the acoustical design of buildings. Students complete a major design project assuming the role of an acoustical consultant. They provide comprehensive acoustical recommendations for all aspects of a major building including room-shaping and finishes, mechanical system noise control, and the design of a speech reinforcement system. They gain experience in field measurement techniques, computer calculations, ray diagramming, and scale modeling. Additional study is pursued through individual research and thesis projects. Examples of previous projects include acoustical scale modeling, the acoustics of the case of an organ, measurements of new indices of acoustical quality in various auditoria, and studies of subjective musical quality. The program is supported by a dedicated laboratory and studio space and three faculty.

ZZ2. Graduate study in acoustics at the Georgia Institute of Technology. Yves H. Berthelot, Jerry H. Ginsberg, Jacek Jarzynski, George M. Rentzepis, and Peter H. Rogers (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

The graduate studies at Georgia Tech span across three colleges, Engineering, Liberal Arts and Sciences, and Architecture and seven schools. Starting with a sequence of three courses on the foundations of physical acoustics, courses in underwater acoustics, atmosphere acoustics, nonlinear acoustics, aeroacoustics, jet noise, noise control architectural acoustics, waves in elastic and viscoelastic media, nonlinear wave propagation, bioacoustics, and vibrations, are available for the formal educational development and preparation for research in acoustics. Opportunities for research across campus are plentiful in such areas as combustion, aerodynamic and helicopter noise, machinery noise, low-noise environmental design, etc. More specifically, at the school of Mechanical Engineering, studies currently in progress are in variational methods in wave propagation and in particular acoustics, light–sound interaction and laser applications to sound detection, fluid–structure interaction and sound propagation, radiation mechanisms, active vibration control, scaling effects, characterization of materials by waves, auditory systems of fish, provide a wealth of graduate thesis topics with a faculty actively engaged in these areas. In addition to the graduate degrees, the institute offers the opportunity for specialized study in acoustics through its multidisciplinary certificate program. Under this program, those who are not as heavily committed to graduate studies in acoustics can be exposed to the field and receive an acoustical engineering or acoustics certificate.

ZZ3. Graduate education in acoustics at Iowa State University. David K. Holger (Department of Engineering Science and Mechanics, 2019 Black Engineering Building, Iowa State University, Ames, IA 50011)

Research and graduate education in a number of areas of acoustics and noise control are carried out in the Department of Engineering Science and Mechanics at Iowa State University. Graduate level courses in theoretical acoustics, signal processing, ultrasonic nondestructive measurements, and mechanical vibrations are offered by the department. Facilities for graduate instruction and research include an anechoic chamber with computer-controlled microphone scanning system, three two-channel FFT analyzers, an ultrasonic nondestructive testing laboratory in the Non-Destructive Evaluation Center, and several thousand square feet of laboratory space. Eight faculty members and more than 20 graduate students are conducting research in spatial transformation of acoustic fields, boundary integral equation methods in acoustics, acoustic intensity methods, ultrasonic transducer characterization, inverse problems in acoustic scattering, and ultrasonic nondestructive evaluation. Descriptions of some typical graduate student programs, selected graduate courses, and specific research projects will be included in the paper.

ZZ4. Graduate program for education in psychological acoustics at the Kyushu Institute of Design. Ryunen Teranishi (Department of Acoustic Design, Kyushu Institute of Design, Shiobaru, Minami-ku, Fukuoka City, 815 Japan)

In Japan, the education of graduate students in universities is done mainly by colloquia and by research performance for their theses or dissertations. Therefore, lecture-type classes are few. In the Institute (a unique university in Japan, which has a department and a course mainly related to acoustics), the above-mentioned conditions are quite similar to other universities. However, the Institute has more acoustical subjects (nine in acoustics and three in musicology, total number of credits = 32) than other universities in Japan. Usually, these have only one or two, as minor subjects, in the electric or electronic engineering course or in the architectual engineering course. Subjects in the Institute program have been carefully selected with the intention of producing acoustic engineers with a deep understanding of nature and the characteristics of humans and the society. Accordingly, almost half of the subjects are related to psychological acoustics, and about one third to musicology. However, the Institute's aim is not to graduate professional musicians or psychologists.

ZZ5. Graduate study in acoustics at Northern Illinois University. Thomas D. Rossing (Department of Physics, Northern Illinois University, DeKalb, IL 60115)

A master's degree is offered in basic or applied physics with specialization in acoustics. Research opportunities in the Acoustics Laboratory include vibrational physics, musical acoustics, psychoacoustics, and environmental noise control. Additional research opportunities exist at two national laboratories and several industrial research laboratories nearby. A doctoral program is in the final stages of development.

ZZ6. Penn State's graduate program in acoustics. Alan D. Stuart (Graduate Program in Acoustics, P.O. Box 30, State College, PA 16804)

The posters will present a comprehensive overview of the graduate program in acoustics at Penn State. In part, the posters will contain the following: (1) descriptions of the acoustics and related courses offered; (2) examples of the facilities available and the research being conducted; (3) listings of graduate theses, completed and in progress; (4) information on special extended education programs—telecommunication and summer—leading to a master's degree in acoustics.

ZZ7. The graduate acoustics education program in the School of Mechanical Engineering at Purdue University. R. J. Bernhard and J. S. Bolton (School of Mechanical Engineering, Purdue University, West Lafayette, IN 47907)

The Acoustics and Noise Control Research Program of the School of Mechanical Engineering at Purdue offers educational and research opportunities at both the undergraduate and graduate level. In addition to undergraduate elective courses in noise control and vibration measurement and control, two graduate engineering acoustics classes are offered within the school. In addition, complementary graduate courses are offered in the areas of digital signal processing, using mechanical vibrations of discretized systems, vibrations of plates and shells, intermediate fluid mechanics, theory and design of control systems, digital control, numerical methods in mechanical engineering, and finite and boundary element methods. Graduate students also have access to a broad spectrum of courses taught in the School of Electrical Engineering and the Department of Physics. Eight faculty members and approximately 20 graduate students participate in the Acoustics and Noise Control Research Program. Most of the research is conducted at the Ray W. Herrick Laboratories. The facilities of the laboratories include an anechoic chamber with a useful working volume of 50 m^3, a reverberation room with a volume of 200 m^3, and a semi-anechoic chamber with a useful volume of 550 m^3. Current research investigations consider: active noise control, active vibration control, structure-borne noise, nonlinear system modeling, numerical noise control design methods, acoustical imaging, modeling of acoustical foams, signature analysis and machinery monitoring, noise source identification, and application of noise control techniques to automotive powertrains and large refrigerant compressors.

ZZ8. The graduate program in neuroscience at Syracuse University. Robert L. Smith and Ronald T. Verrillo (Institute for Sensory Research, Syracuse University, Syracuse, NY 13244-5290)

The Ph.D. program in neuroscience at the Institute for Sensory Research (ISR) of the L.C. Smith College of Engineering provides an excellent opportunity for graduate study and research in many areas of interest to members of ASA. The program is housed at ISR, a multidisciplinary research center devoted to the advanced study of the senses, particularly hearing, touch, and vision. Faculty include several Fellows and Members of ASA: J. J. Zwislocki (auditory psychophysics and biophysics), R. T. Verrillo (vibrotactile psychophysics), R. L. Smith (auditory electrophysiology), E. M. Relkin (auditory electrophysiology), N. B. Slepecky (auditory anatomy), S. J. Bolanowski (cutaneous electrophysiology), and G. A. Gescheider (vibrotactile psychophysics), along with R. B. Barlow (vision), S. C. Chamberlain (anatomy), G. A. Engbretson (vision), D. G. Pelli (vision), and J. Baumgold (neurochemistry). The program of study provides an intensive background in pertinent physical, life, and engineering sciences. Students are directly involved in research work at ISR. They have undergraduate backgrounds in a variety of scientific and engineering areas and generally receive full support through assistantships and fellowships. ISR consists of 27 000 square feet of mostly new, fully computerized, laboratory, office, and auxiliary space and is located on the scenic Skytop campus. A visit is encouraged during the May 1989 ASA meeting.

ZZ9. Graduate acoustics at The University of Texas at Austin. David T. Blackstock, Ilene J. Busch-Vishniac, Mark F. Hamilton, and Elmer L. Hixson (College of Engineering, The University of Texas at Austin, Austin, TX 78712)

The study of acoustics at the University of Texas at Austin is quite broad. It includes physical acoustics, engineering acoustics, underwater acoustics, electroacoustics, noise and vibration control, environmental acoustics, architectural acoustics, psychoacoustics, speech, audiology, linguistics, and musical acoustics. Not surprisingly, the program is spread over several departments. Engineering and physical acoustics is concentrated in the Mechanical Engineering and the Electrical and Computer Engineering Departments, with some activity also in the Physics Department, the Aerospace Engineering and Engineering Mechanics Department, and the Petroleum Engineering Department. Considerable research in hearing and speech is carried out in the Psychology, Linguistics, and Speech Communication Departments. Engineering and physical acoustics is stressed in this paper. Formal course work at the graduate level consists of 1 year of basic physical and engineering acoustics and several advanced one-semester courses, e.g., electromechanical sensors/actuators, nonlinear acoustics, underwater acoustics, ultrasonic engineering, noise control, and wave propagation in continuous media. Research is carried out in all these areas. In the last 8 years, 37 students in the various departments (in the College of Engineering and in Physics) have earned master's degrees with theses in acoustics, and 12 have earned Ph.D. degrees.

ZZ10. Graduate education in underwater acoustics at Tokai University. Minoru Nishimura (Faculty of Marine Science and Technology, Tokai University, Shimizu, 424 Japan)

Research and education in underwater acoustics and its application are indispensable for ocean exploration and development. In 1963, the faculty was established and several subjects on underwater acoustics were included in the undergraduate curriculum. In 1967, the Graduate School of Marine Science and Technology was opened and advanced subjects on underwater acoustics were introduced. Although few students study underwater acoustics, one or two students obtain a master's degree every year, and also five students have taken the degree of Dr. of Engineering in the past 6 years. Studies are mainly directed toward the ultrasonic characteristics of seabed sediment, sound reflection from fish, influence of sound on fish behavior, and nonlinear acoustic characteristics of a focused sound fields. One of the biggest problems is how to make underwater acoustics attractive to the students. The reason that students are less concerned with this field is because it appears unfashionable when compared with electronic studies such as computer science, etc. Another problem is that sea research is physically hard due to the conditions and because it requires much expense and specially designed devices.

THURSDAY AFTERNOON, 17 NOVEMBER 1988 — KAUAI ROOM 2:00 TO 5:58 P.M.

Session AAA. Engineering Acoustic VI and Speech Communication XI: Artificial Intelligence in Acoustics

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Chairman's Introduction—2:00

Invited Papers

2:05

AAA1. AI and signal understanding systems. Raj Reddy (Computer Science Department, Carnegie-Mellon University, Pittsburgh, PA 15213)

Signal interpretation in the presence of noise and uncertainty has always been a difficult problem. In the past 2 decades, a number of application areas have found it necessary to use symbolic knowledge about the situation, context, and language in the form of constraints to reduce exponential search and impove accuracy. This work will provide several examples from the areas of speech, image understanding and target identification, music analysis and recognition, and sensor-based diagnosis in expert systems where symbolic knowledge plays an important role in the interpretation of signals.

2:35

AAA2. Machinery diagnostics. Richard H. Lyon (Department of Mechanical Engineering, Massachusetts Institute of Technology, Cambridge, MA 02139)

Machinery diagnostics means using vibration and/or acoustical signals to determine the performance parameters of the mechanisms and structure of a machine. The signals can be processed in a number of ways to extract the desired information, and commercial systems exist that are able to diagnose in limited but important situations. The purpose of this paper is to review some of the issues that need to be resolved in terms of the knowledge about mechanisms and structures, and signal processing procedures to deal effectively with situations that current systems do not address. These issues have to do with structural complexity and variability, the separation of received information into "source" and "path" components, and the association of faults with particular signatures. Some experiences with the development of customized diagnostic systems are included in the discussion to bring these issues down to earth.

3:00

AAA3. Target recognition in acoustics. G. C. Gaunaurd (Naval Surface Warfare Center, White Oak Laboratory, Code R43, Silver Spring, MD 20903-5000)

The classification/identification of underwater stationary objects can be acoustically achieved by means of passive or active sonar. Moving targets are the "easy case," since they can be identified by their Doppler factor, and will not be discussed here. By means of passive sonar, one merely listens with the best possible receiver to the noises/sounds emitted by the noisy target. The targets are passively "identified" when the spectra of the radiated sound fields they emit are seen to match certain spectral lines known *a priori* for certain desired objects. Identification via active sonar is independent of how noisy the target may be. Here, the echoes returned by the targets when they are insonified by sonar pings are analyzed. These are classical inverse scattering situations in which a returned "signature" in the time or the frequency domain is used to extract physical characteristics of the targets in question. This paper will concentrate on this type of (active) classification technique [i.e., G. Gaunaurd, IEEE J. Ocean. Engr. **OE-12**, 419–422 (1987)], and discuss means to accurately determine size, shape, and composition information about submerged elastic targets from the RESONANCE features present in their monostatic and/or bistatic scattering cross sections. Emphasis will be placed on the resonance scattering techniques (RST) developed for poles in the complex-frequency domain, and their Fourier-related, time-dependent waveforms. The approach is ideally amenable to implementation by expert (AI) systems.

3:20

AAA4. An AI application in acoustic signal processing for target tracking. Douglas C. Dorrough (Advanced Military Avionics Systems, McDonnell Douglas Corporation, Long Beach, CA 90846)

A spectrum of knowledge-based systems has been designed and applied to target acquisition, classification, and tracking. A survey of these leads to the conclusion that more profound techniques need to be developed and applied. Particularly necessary are techniques that match the performance of a highly trained sensor suite operator. Preferably, such a system would function autonomously. A knowledge-based system (called "PING") for autonomously classifying and tracking underwater targets in terms of their acoustic signals as well as their context was designed. PING is based upon earlier work in the foundation of analogical reasoning. The performance of PING was tested and compared with the performance of a population of highly trained sonar operators; the system was also compared with one employing solely the Demster–Shafer evidential reasoning method. In both tests, PING exceeded the performance of the test controls.

3:40

AAA5. SPREX: Speech recognition expert. Riichiro Mizoguchi, Katsuhiko Tsujino, and Osamu Kakusho (I. S. I. R., Osaka University, 8-1 Mihogaoka, Ibaraki, 567 Japan)

Conventional speech recognition systems employ procedural languages such as FORTRAN to encode a large amount of heuristic knowledge, which makes the systems inflexible. SPREX (a SPeech Recognition EXpert)

is implemented as a knowledge-based system that is superior to procedural ones in the use of heuristics. This approach is expected to contribute to the establishment of a new methodology of speech recognition research. SPREX has a knowledge base for feature parameter trajectory reading and the knowledge is encoded in the form of production rules. It has the following three advantages. (1) It can cope with the coarticulation effects. (2) It can be constructed by stepwise development. (3) It is easy to debug. In addition to the recognition system, SPREX also has a powerful environment for knowledge base construction, which consists of five subsystems; a speech database, rule database, rule translator, rule learning module, and interface. The environment helps developers in many respects by employing knowledge engineering techniques. The current version of SPREX is implemented on Symbolics 3620 in Lisp and OPS5e and attains a 95% segmentation rate and an 87% recognition rate for continuous speech 70 s long uttered by six male adults at a speech rate of 6 morae/s. [Work partly supported by a Grant-in-Aid for Scientific Research on Priority Areas.]

4:00–4:10
Break

4:10

AAA6. A score transcription system using musical knowledge. Keiji Hirata (NTT Software Laboratories, 3-9-11 Midoricho, Musashino, 180 Japan) and Tatsuya Aoyagi (The University of Electro-Communications, 1-5-1 Chofugaoka, Chofu, 182 Japan)

Automatic score transcription systems have two problems. (1) How can a machine make a score from the results of an FFT? Particularly in the case of polyphonic sound, it is hard to decide whether peaks are the overtones or the fundamentals of the actually played notes. (2) How can a machine reduce the amount of FFT computation? Usually the FFT is performed while shifting the FFT window by a small amount of time. This results in a great deal of computing time consumed by the FFT. To overcome the above problems, musical knowledge is used. Assuming that an input signal is a musical sound, a system can eliminate unnecessary possibilities and wasteful computation. For instance, a system uses musical knowledge about reasonable notes on current scales. Further, the parameters for FFT computation can be optimized with musical knowledge. The musical knowledge decides the degree of the musical importance of data; according to this degree, this system can adjust the time and frequency of the FFT computation. For example, since the onset of a note is more important than its decaying part for the score transcription, the system focuses its computing power onto the onset part.

4:30

AAA7. Spatialized computation in sound and music. Bernard Mont-Reynaud (Center for Computer Research in Music and Acoustics, Stanford University, Stanford, CA 94305)

The PRISM system (for Pattern Recognition In Sound and Music) is the acoustic front-end of an experimental research environment that attempts to imitate qualitative aspects of human hearing; examples include polyphonic music recordings. This paper focuses on the acoustic representation used in PRISM, an alternative to spectrograms well suited for perception work; the method used to compute the representation is described elsewhere. An ideal representation for analysis offers mathematical elegance, ease of computation, psychological plausibility, and perceptual immediacy. Linear time suits most purposes, but psychoacoustic sensitivity studies suggest a quasilogarithmic frequency scale; and psychological, musical, and mathematical reasons lead to using $\log(f)$ for the pitch scale. With the latter choice, pitch invariance maps to translation invariance; and musical intervals, scales, and harmonic series are patterns revealed by straightforward convolution, or by the naked eye. This leads to a promising family of algorithms for polyphonic pitch detection and other fundamental acoustic analysis tasks. Intuitive user interfaces and visualization tools are obtained from the representation. Last but not least, the algorithms map readily to most massively parallel architectures. [Work supported by NSF.]

4:50

AAA8. A system for knowledge-based architectural acoustics. Masuzo Yanagida (Communications Research Laboratory, 4-2-1 Nukui-kitamachi, Koganei, 184 Japan), Takayuki Hidaka (Takenaka Technical Research Laboratory, 2-5-14 Minami-suna, Kohtoh-ku, Tokyo, 136 Japan), and Takahira Yamaguchi (ISIR, Osaka University, 8-1 Mihogaoka, Ibaraki, 567 Japan)

The concept of knowledge-based architectural acoustics is proposed showing a prototype of an intelligent system for room acoustic design as a practical example. The system is composed of two major subsystems, an acoustic field simulator and an expert system for room acoustics. The principal objective of the system is to

assist acoustic designing. This system can also be used for training acoustic designers and for evaluating both design policies and design parameters to represent the physical characteristics and subjective aspects of sound fields, such as spaciousness and definition. The field simulator is utilized effectively by refining the calculation algorithms and checking whether the performance coincides with actual fields. The validity of the design parameters is examined by listening to the output sounds of the field simulator. Some examples of the system's performance on rooms with simple shapes are shown, and the principle of evaluating subjective design parameters is indicated.

Contributed Papers

5:10

AAA9. Prognosis and diagnosis using acoustics. David Antonelli and Caroline Fu (Research and Engineering Division, Boeing Aerospace, P. O. Box 3999, Mail-Stop 82-58, Seattle, WA 98124-2499)

Present methods of prognostics and diagnostics on engines, especially aircraft engines, involve performance monitoring and wrap testing using built-in test (BIT) and built-in test equipment (BITE). Performance monitoring is used primarily for measuring liquid flow, temperatures, and current flow. Wrap testing is used primarily for testing electrical circuits. If the hypothesis holds that vibration and/or acoustical signals can be interpreted to detect changes in a mechanical operation, this will add another dimension to engine monitoring, diagnostics, and prognostics. Abnormalities in acoustic signals emitted by worn mechanical parts could be detected and warn the operator of imminent failure. This paper describes how the SPIRE research tool could be modified and integrated with AI techniques to interpret acoustic patterns to predict engine failure.

5:22

AAA10. An automated vibration identification system for complex machinery installations. Ph. Esparcieux (Direction des Constructions et Armes Navales de Tolon-Cerdan-BP77, 83800 Toulon-Naval, France)

Excessive vibrational levels on complex machinery installation, like a ship, may cause considerable annoyance in both the environment and the installation itself. Because noise and vibration specialists are not always available in all situations, an attempt has been made to proceed to a purely automated identification procedure. Attention has been focused on situations in which critical levels are established on crucial locations of the structure (the so-called "monitoring points"), rather than on the sources themselves. A vibration monitoring system that involves a careful transducer arrangement and an extensive data acquisition system is fitted on the installation. Prior knowledge of mounted equipment and dynamic characteristics of structural elements is necessary to constitute a description data base that is referred to by an expert system that supervises the entire procedure, including data acquisition and specific signal processing. Then, after a preselection of the suspected sources, the supervisor system controls a more detailed scrutinization that yields the final diagnosis of the fault of the installation. The paper presents a fully automated identification system that works on an installation fitted with 18 rotating machineries.

5:34

AAA11. Inductive generation of phoneme recognition knowledge for a continuous speech recognition system SPREX. Katsuhiko Tsujino, Riichiro Mizoguchi, and Osamu Kakusho (I. S. I. R., Osaka University, 8-1 Mihogaoka, Ibaraki, 567 Japan)

A continuous speech recognition expert system named SPREX has been constructed. It simulates the behavior of human experts who can recognize speech by inspecting the trajectories of feature parameters such as formant and power [R. Mizoguchi *et al.*, Proc. ICASSP '86 2, 1221–1224 (1986)]. In order to construct a speaker-independent knowledge base, it is necessary to learn by concrete speech samples. ARIS (adaptive rule induction system) is developed as an inductive learning subsystem for SPREX. It has the following three characteristics that are also interesting from a knowledge of engineering point of view: (1) It can generate new attributes for separating the training samples efficiently; (2) it can cope with the noisy data problem by using heuristic generalization operators; and (3) it can deal with numerical attributes as well as symbolic ones. The latest version of a knowledge base generated by ARIS recognizes 82% of consonants and 95% of vowels. Some of these performances exceed that of the rules written by human experts. [Work partly supported by a Grant-in-Aid for Scientific Research on Priority Areas.]

5:46

AAA12. Stress-induced speech and speech recognition. John W. Gordon (Boeing Aerospace Company, P. O. Box 3999, Mail-Stop 82-58, Seattle, WA 98124)

The performance of speaker-dependent speech recognizers typically degrades when the speech input varies from its trained norm, particularly when the speaker is placed under stress. To evaluate such performance, a data base of isolated words, collected with the speakers under various simulated environmental conditions or emotional states, was used to train and test two commercial recognizers. For comparison purposes, the recognition performance of human subjects on the same data base was also obtained. The speech was transcribed, and the acoustic properties of the phonetic tokens from various speaking styles were analyzed; this facilitated the investigation of correlations between degradations in performance and speech variabilities. Results indicate the importance of direct representation of speech knowledge in speech recognition systems that are used in situations where changes in speech properties are to be expected.

Session BBB. Structural Acoustics and Vibration VI: Acoustic Imaging of Vibrating Structures

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Chairman's Introduction—2:00

Invited Papers

2:05

BBB1. Acoustic holography for wideband, arbitrarily shaped noise sources. J. D. Maynard (The Pennsylvania State University, 104 Davey Laboratory, University Park, PA 16802)

Acoustic holography has been shown to be a powerful technique for the study of sound radiation from complex structures. In its most rudimentary form, acoustic holography involves the measurement of a single frequency of sound throughout a spatial surface that corresponds to the level surface of a separable coordinate system. Most holography systems are limited to this basic form because of data acquisition and processing limitations. However, a high-speed microphone array data acquisition system with an on-line array processor and new data processing algorithms that permit acoustic holographic reconstructions of wideband noise sources with arbitrarily shaped surfaces has been developed. The data acquisition system is now described in detail in the literature. [Maynard *et al.*, J. Acoust. Soc. Am. **78**, 1395–1413 (1985); W. A. Veronesi and J. D. Maynard, J. Acoust. Soc. Am. **81**, 1307–1322 (1987)]. In this paper, holography for odd-shaped surfaces, i.e., surfaces that do not coincide with level surfaces of separable coordinate systems, will be emphasized. Current algorithms involve finite element techniques and singular value decomposition. [Work supported by the Office of Naval Research.]

2:25

BBB2. Acoustic source imaging using numerical models. Robert J. Bernhard and B. K. Gardner (School of Mechanical Engineering, Purdue University, West Lafayette, IN 47907)

An acoustic imaging method is formulated based on boundary element numerical approximations to the Helmholtz integral equation. A geometric model of the source and any important environmental features such as scatterers or enclosures are used. The numerical model quantifies the interrelationship of various parts of the source and the interaction of the source and its environment. Known boundary condition data and field pressure data are used to specify the problem. The unknown boundary conditions of the acoustic source are solved. The boundary solution describes the acoustic source. Using the boundary solution, other desired information can be numerically computed to further describe the interaction of the source with its environment. The method is intended to complement nearfield acoustic holography for sources of general shape and for applications in reactive acoustic fields or where scatterers are present. The method is more computationally intensive than holographic methods but is more versatile and requires less measured data. Results will be shown for the method applied to sources in enclosures. In addition, the sensitivity of the method to random and bias errors will be illustrated. [Work supported by NASA Grant NAG1-58.]

2:45

BBB3. Current advances in cylindrical nearfield acoustical holography. Earl G. Williams (Naval Research Laboratory, Code 5137, Washington, DC 20375-5000)

Probably no other technique available to the experimenter provides such an exhaustive exposition of the structural acoustics of fluid loaded vibrating cylinders in such a short time, as does broadband nearfield acoustical holography. In a single experiment consisting of the measurement of the pulsed pressure on a 2-D cylindrical contour, the complete 3-D acoustic fields outside the cylinder are obtained over a frequency range of 2 octaves. Recent results on high-mode density cylinders will be shown, including the validation of the broadband results by comparison of the holographic, radial velocity reconstruction with direct measurement from a surface mounted accelerometer. Because of the volume of data provided by this technique, computer graphics are used to help expose physical phenomena. A videotape displaying simultaneously the oscillation of the surface velocity and nearfield pressure fields resulting from a shaker driven cylinder, as functions of frequency, will be shown. Current work will be discussed in boundary element implementation of holography using the Helmholtz integral, for application to axisymmetric, capped cylinder structures.

3:05

BBB4. Backward reconstruction in acoustical holography based on iterative inversion techniques. M. R. Bai and A. L. Pate (Department of Engineering Science and Mechanics, Iowa State University, Ames, IA 50011)

Reconstruction of a source surface motion from acoustic holographic data presents computational difficulties because the problem is ill-posed as is well known in the literature. In order to deal with these difficulties, a method based on a recursive algorithm was developed. In this method, the inverse problem is converted to a well-posed forward propagation problem. An initial guess regarding the source images is required to activate the iterative inversion method. Then, the tentative image is forward propagated to the hologram plane and the residue is determined. Next, a feedback operator is used to process the residue by which the image is updated. Two types of feedback operators were investigated: (1) Wiener type, suboptimal operator, and (2) dynamic, optimal operator (designed subject to minimum mean-square error optimization criteria). Both iteration methods provided satisfactory convergence. In addition, these methods were found to be relatively insensitive to the choice of the initial guess and the noise parameters used in feedback operators. The efficiency of the iteration process is greatly enhanced by 2-D FFT. Numerical and experimental results obtained by using iterative inversion techniques will be presented. The iterative inversion techniques will be compared with conventional inverse filtering methods. In addition, performance of the iterative inversion methods when used in connection with the nearfield acoustic holography technique will be discussed. [Work supported by NSF.]

3:25

BBB5. The use of nearfield acoustic holography in noise control applications. U. D. Dietschi (Clark Laboratory Services, 821 East Front Street, Buchanan, MI 49107) and J. S. Bolton (Ray W. Herrick Laboratories, School of Mechanical Engineering, Purdue University, West Lafayette, IN 47907)

Nearfield acoustical holography (NAH) allows a radiated sound field to be reconstructed from complex pressure measurements made on a plane in the nearfield of a source. In this way, source normal surface velocities and surface normal intensities may be determined. The latter quantity may be used to locate radiating acoustical sources and so guide noise control treatments. However, the availability of NAH systems has been limited by their cost and complexity. In addition, all current systems (with the exception of that developed by Hald and Ginn) require explicit or implicit knowledge of the forcing function that results in the sound radiation in order to derive the complex pressure field; this information is not normally available to the noise control engineer. At the Herrick Laboratories, a system addressing both of these concerns has been built using instrumentation commonly available in acoustics laboratories. It may be considered to be a single reference implementation of the cross-spectral method developed by Hald and Ginn. Measurements have been made of both ideal sources and a small appliance. The results have been shown to compare well with theoretical results and with conventional intensity measurements.

3:45

BBB6. Acoustical imaging using the propagation law of mutual intensity. Sadayuki Ueha (Research Laboratory of Precision Machinery and Electronics, Tokyo Institute of Technology, 4259 Nagatsuta, Midori-ku, Yokohama, 227 Japan)

Vibrating objects are successfully imaged using acoustical holography where the radiated sound is used both for the object and the reference signals [Ueha *et al.*, Appl. Opt. **14**, 1478–1479 (1975)]. The same technique can be used both for partially coherent and incoherent objects. But the characteristics of the reconstructed images depend on the degree of coherence of the objects under study [Ueha *et al.*, Opt. Commun. **18**, 488–491 (1976)]. In order to treat reconstructed images qualitatively, this imaging technique is reviewed from the viewpoint of the propagation law of mutual intensity. In addition, the effects of the spectral bandwidth and signal averaging time upon the reconstructed images are discussed.

4:05

BBB7. A noise source detection system using nearfield acoustical holography. Takuro Hayashi (Toshiba Research and Development Center, 4-1 Ukishimacho, Kawasaki-ku, Kawasaki, 210 Japan)

A noise source detection system based on nearfield acoustical holography has been developed. This system consists mainly of a measuring section, a hologram section, and a reconstruction section. In the measuring section, acoustic waves radiated from sound sources are sampled simultaneously with both a scanning microphone in the measurement plane and a spatially fixed reference microphone. The hologram section obtains the hologram data set from the cross spectra between the two microphone signals at each sampling point. The reconstruction section processes the hologram data to reconstruct the acoustic wave field radiated from the original sound sources. In addition, three ways for acoustic image reconstruction, using evanescent wave components or plane-wave components, or both, can be selected at this reconstruction stage. Sound source identification tests for sound sources, such as loudspeakers, were carried out using this system. The results showed that a good spatial resolution could be obtained, and that the sound source locations could be identified satisfactorily.

4:17

BBB8. Acoustical imaging and farfield radiation patterns in air and water using STSF. Joseph A. Clark (Catholic University of America, Washington, DC 20064)

The spatial transformation of sound fields (STSF) technique involves a planar scanning across the source under investigation. From each scan position, the cross spectrum is measured to every one of a set of references and, further, the cross spectrum is measured between each pair of the references. From the measured cross spectra, any descriptor of the nearfield (pressure, particle velocity, intensity, etc.) can be investigated by means of nearfield acoustic holography, while the more distant field can be determined by application of Helmholtz's integral equation. The cross spectrum formulation implemented in the STSF technique has the advantage of putting no restrictions on the coherence of the sound field. Only a certain stationarity of the source during the measurement procedure has to be assumed. This paper describes the use of the STSF measurement method in an investigation of radiation from dipoles both in air and in water. The discussion will briefly review the principle of the STSF method and then will present pressure, velocity, and active, and reactive intensity plots calculated in the plane of the dipole sources from data measured in the acoustic nearfield. Farfield radiation patterns calculated from the same nearfield measurement data will also be shown and compared with sound-pressure levels measured directly in the farfield. Some factors influencing the accuracy of STSF measurements of farfield radiated noise will be identified.

4:29

BBB9. Visualization of the aerial ultrasound field in the nearfield of a transducer. Tetsuro Otsuka, Takashi Shiode, and Koichiro Seya (College of Industrial Technology, Nihon University, Narashino, 275 Japan)

High-power aerial ultrasound sources having a stepped circular vibrating plate were designed for frequencies of 20 and 28 kHz. The diameter of the vibrating plate was about 5 λ (λ is the wavelength in air). The purpose of this sound source was to make a piston sound source having a diameter much larger than the wavelength so that the single beam directional pattern could be obtained on the center axis of the sound field. To investigate the characteristics of the sound source, the imaging of the ultrasound field radiated from the sound source was experimentally performed. The signal from a scanning microphone in the sound field was introduced to a lock-in amplifier and was compared with the reference signal. The output signals of the phase and the amplitude from the lock-in amplifier were linked to a microcomputer and a work station. The ultrasound field was visualized on a CRT display with 256 different colors. The characteristics of the propagating ultrasound from the stepped circular vibrating plate are discussed.

4:41

BBB10. Boundary element implementation of generalized nearfield acoustic holography for an axisymmetric geometry. Giorgio Borgiotti (Department of Electrical Engineering, George Washington University, Washington, DC 20037), Angie Sarkissian, Luise Schuetz, and Earl Williams (Naval Research Laboratory, Code 5130, Washington, DC 20375-5000)

Generalized nearfield acoustic holography (GENAH), which allows a three-dimensional reconstruction of the sound-pressure field from two-dimensional measurements of the pressure in the nearfield of a vibrating structure, is extended to nonseparable coordinate systems, in particular, a general axisymmetric structure. This technique allows a high-resolution imaging of the normal velocity at all points on the surface of the body. The Helmholtz integral equation formulation is used, with singular value decomposition of the matrix to treat the presence of evanescent waves in the nearfield. Symmetry principles are implemented to reduce computation times for the axisymmetric geometry.

4:53

BBB11. Transient noise source location via computational and experimental surface acoustic intensity methods with an application to a punch press. D. H. Chun and Thomas H. Hodgson (Center for Sound and Vibration, Mechanical and Aerospace Engineering Department, North Carolina State University, Raleigh, NC 27695-7910)

This paper outlines a recent extension of the boundary element computational method to the case of transient acoustic energy flow from a complex mechanical structure. Results are compared with experiment by use of the fiber-optic surface acoustic intensity probe. The computational method presented combines an explicit and implicit method to reduce the numerical instability inherent in the method and to increase the efficiency of the solution for such problems. Also, a recent improvement of the fiber-optic surface acoustic intensity probe will be discussed to illustrate the validity of the experimental method, which eliminates the phase error in higher frequency measurements and increases the sensitivity and linearity. The application of these methods on a small punch press will be described to demonstrate the capability and accuracy of the methods. Importance of nonresonant frequency modes in the transient mechanism will also be discussed by analyzing the simulated frequency responses where resonant components have been eliminated.

5:05

BBB12. Visualization of acoustic scattering from submerged objects. Charles F. Gaumond, Lawrence C. Schuette (Codes 5132 and 5133, Naval Research Laboratory, Washington, DC 20375-5000), and Joseph G. Gatto (Sachs Freeman Associates, Landover, MD 20785)

The scattering of harmonic plane waves from rigid spheres, rigid cylinders, and cylindrical shells was visualized using normal mode series solutions displayed on an IRIS-4D color graphic super-workstation. The temporal behavior of the pressure and the instantaneous energy density were studied for $ka < 10$ in a square region 150×150 ka units in size. Particular attention was paid to processes that generate phase shifts in the forward direction. A video presentation of the temporal behavior will be shown.

THURSDAY AFTERNOON, 17 NOVEMBER 1988 MAUI ROOM, 2:00 TO 4:12 P.M.

Session CCC. Underwater Acoustics VII: Shallow Water Acoustics II

Robert F. Gragg, Cochairman
Code 5160
Naval Research Laboratory
Washington, DC 20375

Toshiaki Kikuchi, Cochairman
Department of Applied Physics
National Defense Academy
1-10-20 Hashirimizu
Yokosuka, 239 Japan

Contributed Papers

2:00

CCC1. Stability of marching methods applied to elliptic models of underwater sound propagation. G. H. Knightly and D. F. St. Mary (Center for Applied Mathematics and Mathematical Computation, Department of Mathematics, University of Massachusetts, Amherst, MA 01003)

Recently [G. H. Knightly and D. F. St. Mary, in *Computational Acoustics: Wave Propagation*, edited by D. Lee, R. L. Sternberg, and M. H. Schultz (Elsevier, Amsterdam, 1988), pp. 397–407] an investigation of marching methods for the farfield elliptic equation of underwater sound propagation was begun. The elliptic model is obtained on transforming the Helmholtz equation, written in terms of the pressure p, by taking $p(r,z) = w(r,z)H_0^{(1)}(k_0r)$, and using the farfield properties of the Hankel function $H_0^{(1)}$. Using a marching scheme to solve the resulting equation assumes that this elliptic problem has been cast as an initial value problem, and of course initial value problems for elliptic partial differential equations are in general not well posed and yield unstable numerical schemes. But such methods may be appropriate under certain restrictions on the parameters. Two marching schemes for computing solutions of the model are demonstrated and stability for these schemes is discussed.

2:12

CCC2. A normal-mode model for calculating bistatic reverberation in shallow water. Dale D. Ellis, Anna M. Crawford[a] (Defense Research Establishment Atlantic, P.O. Box 1012, Dartmouth, Nova Scotia B2Y 3Z7, Canada), and Stephen N. Wolf (Code 5160, Naval Research Laboratory, Washington, DC 20375)

A model for calculating shallow-water reverberation in bistatic geometries has been developed using normal modes and ray-mode analogies. The acoustic propagation from the source to the scattering area, and from the scattering area to the receiver is described using normal modes. The scattering at the sea–surface or sea–bottom interface is described by a ray-type scattering function, assumed to be known. Ray-mode analogies are used to relate the modal wavenumbers to vertical angles, and to obtain an expression for the scattering area. Travel times are determined from the modal group velocities. The theoretical formulation applies to fairly general environments; a model for a Pekeris-type environment has been implemented for bottom reverberation due to short pulses. Results will be compared with measured reverberation from an explosive source received on a horizontal line array. [a] Currently a student in the Department of Physics, Dalhousie University, Halifax, Nova Scotia B3H 3J5, Canada.

2:24

CCC3. Scattering in a shallow water waveguide. Gary S. Sammelmann and Roger H. Hackman (Physical Acoustics Branch, Naval Coastal Systems Center, Panama City, FL 32407)

The bistatic scattering from an elastic target in a shallow-water waveguide with a rigid bottom was discussed at a previous meeting of the society [G. S. Sammelmann and R. H. Hackman, J. Acoust. Soc. Am. Suppl. 1 82, S49 (1987)]. One emphasis of this work was the effect of modal dispersion and the strong elastic signature of the flexural resonances on the scattered waveform. Here, an extension of the model to realistic bottoms is given. In particular, the effects of bottom structure on the time domain structure of the scattered signal are discussed and the generation and propagation of interface waves and the scattering of such waves by a target in the waveguide are considered.

2:36

CCC4. Ray analysis near the critical angle. C. T. Tindle and N.G. Plumpton (Physics Department, University of Auckland, Auckland, New Zealand)

The sound field due to spherical waves from a point source reflected from the interface between two fluids is re-examined. The conventional ray theory description in terms of a specular ray and a lateral wave is satisfactory at both short and long ranges but fails at intermediate ranges when the specular ray is near the critical angle. Ray theory with beam displacement leads to the formation of a caustic and gives good results beyond the caustic range but fails at shorter ranges. In the present work, ray theory with beam displacement is extended to allow for rays with complex wavenumber by associating rays with saddle points of a complex integrand. The field is calculated by the method of steepest descents and compared with the exact integral result. The field is smooth and continuous through both critical angle and caustic and in good agreement with the exact result. The method is based on eigenrays and should enable ray calculations to be extended to accommodate critical angles. The description of beam displacement caustics is similar to earlier treatments of refraction caustics in terms of complex rays.

2:48

CCC5. Directional spectral and particle motion measurements of underwater seismic noise. Dean Goodman and Tokuo Yamamoto (Geo-Acoustics Laboratory, Rosenstiel School of Marine and Atmospheric Science, University of Miami, 4600 Rickenbacker Causeway, Miami, FL 33149)

Observations of underwater seismic noise were measured using arrays of ocean bottom seismometers (OBS) deployed in shallow waters on the continental shelf off New Jersey. To reduce uncorrelated noise on the seafloor, instruments were buried to 0.5-m depth with the aid of an underwater hydraulic pump. Array geometry and dimensions were tuned to measure microseismic noise in the period band of 2–6 s. Directional spectrums measured by maximizing cross spectrums between array elements for a given wavenumber indicate that slow microseismic waves having an apparent velocity of 200–300 m/s were traveling out of the east. Phase velocities determined from normal mode analysis agree closely with measured phase velocities. From particle motion analysis for single OBS elements, microseismic noise is shown to exhibit retrograde motion at the seafloor. These experiment results indicate that microseismic noise is due to Scholte-type interface waves. [Work supported by ONR, code OA.]

3:00

CCC6. High-resolution remote sensing of geoacoustic properties of the seabed by a buried OBS. Tokuo Yamamoto, Mohsen Badiey, Mark Trevorrow, and Altan Turgut (Geo-Acoustics Laboratory, Rosenstiel School of Marine and Atmospheric Science, University of Miami, 4600 Rickenbacker Causeway, Miami, FL 33149)

An entirely new geophysical inverse theory has been developed to remotely measure the shear modulus versus depth profile of the seabed sediments using the gravity wave induced seabed motion [Yamamoto and Torii, Geophys. J. R. Astron. Soc. **85**, 413–431 (1986)]. From several experiments conducted in shallow water (4–135 m), shear modulus profiles were extracted down to 200-m sediment depths, and at a resolution of 1 m [e.g., Trevorrow *et al.*, Geophys. J. R. Astron. Soc. (to be published)]. There is a unique relation between the shear modulus and the porosity of sediment at a given overburden pressure. The porosity versus sediment depth profile is uniquely determined from the shear modulus profile using this relation. Finally, V_p, V_s, Q_p^{-1}, Q_s^{-1} versus sediment depth profiles are obtained through the Biot–Yamamoto theory [Yamamoto and Turgut, J. Acoust. Soc. Am. **83**, 1744–1751 (1988)]. Comparisons between the BSMP remote sensing results and borehole data at five AMCOR and other geophysical borehole sites show excellent agreement. Prediction of transmission loss using BSMP geoacoustic data agrees very well with experiments conducted at the New Jersey Shelf. [Work supported by ONR Code 11250A.]

3:12

CCC7. Seismoacoustic measurements within the sedimentary subbottom of the Ligurian Shelf. T. G. Muir, T. Akal, G. Guidi, and E. Michelozzi (SACLANT Undersea Research Centre, Viale San Bartolomeo 400, 19026 La Spezia, Italy)

A three-axis seismometer was deployed at a depth of 6 m in a mud sediment by first making a 20-cm-diam hole in the bottom, using a bottom-mounted "drilling" machine, inserting the seismometer, and then removing all machine components, allowing sediment to cave in around the sensor. An identical seismometer was deployed on the sediment/water interface. Data were acquired with both ambient noise and explosive shot sources detonated on the bottom as a function of range to 3 km in a water column 16 m deep. While the bottomed seismometer data showed the existence of Sholte-type interface waves present on both horizontal (radial) and vertical geophones, the subbottom seismometer data show signals predominantly on the vertical geophone. The results are analyzed and modeled with the SAFARI code.

3:24

CCC8. Velocity and attenuation from shallow-water experiments. Robert D. Stoll (Lamont-Doherty Geological Observatory, Palisades, NY 10964)

During the summer of 1987 a series of shallow-water experiments were performed for the purpose of determining the seismoacoustic properties of the unlithified sediments near the seafloor. These experiments, which utilized a small explosive source and a string of receivers on the seafloor, were designed to maximize the excitation of shear and interface waves. By using two-axis, gimballed geophones spaced at 5-m intervals along a cable, a robust set of data defining motion in a vertical plane parallel to the direction of propagation was obtained. By analyzing the dispersion of surface waves and the amplitude and travel time of diving shear waves, the data were inverted to obtain shear and dilatational wave velocity and the attenuation of shear waves as a function of depth. One data set for a thick sand deposit in New York Harbor showed strong gradients in velocity and attenuation near the water–sediment interface in good agreement with a theoretical model which takes into account the rapid change in effective stress (overburden pressure) in the near-bottom sediments. [Work sponsored by ONR, code 1125 QA.]

3:36

CCC9. A modal/WKB inversion method for determining sound-speed profiles in the ocean and ocean bottom. Kevin D. Casey (MIT/WHOI Joint Program in Oceanography/Oceanographic Engineering, Woods Hole, MA 02543) and George V. Frisk (Department of Ocean Engineering, Woods Hole Oceanographic Institute, Woods Hole, MA 02543)

Two approaches to determining sound-speed profiles in the ocean and ocean bottom using measured acoustic modal eigenvalues are examined. Both methods use measured eigenvalues and mode-dependent assumed values of the WKB phase integral as input data and use the WKB phase integral as a starting point for relating the index of refraction to depth. Inversion method 1 is restricted to monotonic or symmetric sound-speed profiles and requires a measurement of the sound speed at one depth to convert the index of refraction profile to a sound-speed profile. Inversion method 2 assumes that the sound speed at the ocean surface and the minimum sound speed in the profile are known and is applicable to monotonic profiles and to general single duct sound-speed profiles. For asymmetric profiles, inversion method 2 gives the depth difference between two points of equal sound speed in the portion of the profile having two turning points, and in the remainder of the profile it gives sound speed versus depth directly. A numerical implementation of the methods is demonstrated using idealized ocean sound-speed profiles. The two methods are used also to determine the sediment sound-speed profiles in two shallow water waveguide models, and inversion method 1 is used to find the sediment sound-speed profile using data from an experiment perfomed in the Gulf of Mexico. [Work supported by ONR.]

3:48

CCC10. Development of a compact apparatus to measure both the complex bulk modulus and the complex mass density of a fluid contained in a rigid porous solid. Steven D. Grant, Steven R. Baker, and Oscar B. Wilson (Department of Physics, Code 61Ba, Naval Postgraduate School, Monterey, CA 93943)

Two methods for determining the complex bulk modulus B and the complex fluid mass density ρ of a fluid contained in a rigid porous solid are being investigated. The fluid-filled solid is contained within a small ($\ll \lambda$) cylindrical cavity capped on each end by identical transducers. One method, suitable for acoustically "hard" transducers, is based upon a direct application of Newton's second law. Here, B and ρ are extracted from the pressure at each transducer which results from its known velocity when the transducers are caused to oscillate, "in-phase" and "out-of-phase," respectively. The other method, suitable for acoustically "soft" transducers, is based upon electroacoustic network theory. The B and ρ are obtained from the input electrical impedance of each transducer when they are wired in parallel "in-phase" and "out-of-phase," respectively. A description of the apparatus and preliminary experimental results will be presented. [Work sponsored by NRL-USRD.]

4:00

CCC11. Effects of surface bubble layers on coastal acoustic tomography transmissions. Iwao Nakano and James. F. Lynch (Japan Marine Science and Technology Center, Yokosuka 237, Japan and Woods Hole Oceanographic Institution, Woods Hole, MA 02543)

Recent work by Lynch, Miller, and Chu has shown that one can obtain surface wave frequency-directional spectra with low-frequency ($\sim$225 Hz) acoustic tomography in coastal regions [Lynch *et al.*, J. Geophys. Res. **92**, 6869–6885 (1987)] using the travel-time fluctuations of the acoustic multipaths as data. However, no account was taken in this work to estimate the effects on the acoustics of the near surface bubble layer produced by the surface waves. This bubble layer reduces the sound speed near the surface and produces two effects: First, a travel-time delay is produced, i.e., the bubble layer "biases" the travel time estimate; and second, the interaction angle of the ray with the surface is steepened. The dependence of both of these effects on wind speed and ray angle are studied for a simple waveguide example. A comparison of the magnitude of the bubble layer effects with the surface scattering effects is also made.

THURSDAY AFTERNOON, 17 NOVEMBER 1988

IAO NEEDLE/AKAKA FALLS ROOM, 3:00 P.M.

Meeting of Accredited Standards Committee S3 on Bioacoustics

to be held jointly with the

Technical Advisory Group (TAG) Meeting for ISO/TC 43 Acoustics, IEC/TC 29 Electroacoustics, IEC/TC 87 Ultrasonics, and ISO/TC 108/SC4 Human Exposure to Mechanical Vibration and Shock

L. A. Wilber, Chairman S3

422 Skokie Boulevard, Wilmette, Illinois 60091

Standards Committee S3 on Bioacoustics. The current status of standards under preparation will be discussed. In addition to those topics of interest including hearing conservation, noise, dosimeters, hearing aids, etc., consideration will be given to new standards which might be needed over the next few years. Open discussion of Committee reports is encouraged.

The international activities in ISO/TC 43 Acoustics, IEC/TC 29 Electroacoustics, IEC/TC 87 Ultrasonics, and ISO/TC 108/SC4 Human Exposure to Mechanical Vibration and Shock will be discussed. The Chairs of the TAGs for ISO/TC 43 (H. E. von Gierke), IEC/TC 29 (V. Nedzelnitsky), IEC/TC 87 (P. D. Edmonds), and ISO/TC 108/SC4 (H. E. von Gierke), will report on current activities of these Technical Committees and Subcommittees.

Reports will be given on the meetings of ISO/TC 43 and IEC/TC 29 (taking place in Toronto, Canada in October 1988), the meeting of IEC/TC 87, in Philadelphia, PA from 24–28 October 1988, and on the meeting of ISO/TC 108/SC4, in Canton, People's Republic of China, from 5–9 September 1988.

THURSDAY EVENING, 17 NOVEMBER 1988

6:30 Social Hour (cash bar)—Lobby Foyer, Second Floor
7:30 Banquet—Hawaii Ballroom

Presentation of Acoustical Society of America Awards

Pioneers of Underwater Acoustics Medal to Robert J. Urick
Silver Medal in Noise to William J. Galloway
Silver Medal in Physical Acoustics to Mack A. Breazeale

Presentation of Acoustical Society of Japan Awards

Sato Medal to Minoru Takahashi, Takashi Kuribayashi, Kin-ichiro Asami, Takashi Enokida, Hareo Hamada, and Tanetoshi Miura
Sato Medal to Shigeo Ohtsuki, Motoyoshi Okujima, and Motonao Tanaka

Session DDD. Musical Acoustics V: East and West

H. John Sathoff, Cochairman
Physics Department
Bradley University
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Junji Takahashi, Cochairman
Laboratory for Musical Acoustics
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Contributed Papers

8:00

DDD1. Modal scales of the Dan Tranh. Douglas H. Keefe, Phong Nguyen (Systematic Musicology and Ethnomusicology Programs, School of Music, DN-10, University of Washington, Seattle, WA 98195), and Edward M. Burns (Department of Speech and Hearing Sciences, University of Washington, Seattle, WA 98195)

The Dan Tranh is a Vietnamese 17-string zither tuned to pentatonic modal scales. Each mode encompasses a given tuning and a specific mental expression or modal nuance, and ornamentation using pitch bends is an important embellishment. The Dan Tranh was tuned entirely by ear, and the repetition rate of the lowest-pitched string varied from G3 to E4 in data obtained over a 6-month period from a single artist (P.N.). The octave stretch was approximately 13 cents over 3 octaves with most of the increase in the upper octave, and was subtracted out to define a mean modal tuning. The intervallic distances and standard deviations in cents relative to the tonic of each mode are:

$$\text{Bac} = [0, 188 \pm 7, 500 \pm 9, 698 \pm 7, 891 \pm 4],$$

$$\text{Ho Mai Nhi} = [0, 166 \pm 10, 513 \pm 5, 695 \pm 9, 859 \pm 9],$$

$$\text{Oan} = [0, 338 \pm 9, 503 \pm 6, 699 \pm 6, 1026 \pm 15],$$

and

$$\text{Vong Co} = [0, 375 \pm 13, 519 \pm 8, 704 \pm 10, 866 \pm 6].$$

The octave stretch and tuning variability are smaller than comparable data obtained in Western and other music cultures. The Vong Co mode strongly and the Ho Mai Nhi mode weakly violate coherency as defined by Balzano (1980), but two additional pitches in the Vong Co mode obtained through ornamentation restore coherency. The artist's ability to categorize isolated melodic intervals in a laboratory context will be discussed. [Work partially supported by NIH.]

8:15

DDD2. Scalar correspondence with the overtone series. Bennett S. Blum (Department of Critical Studies, Massachusetts College of Art, Boston, MA 02115)

It is frequently stated that the overtone series produces the notes (starting on C) C, G, E, (flat) B♭, D, (flat) F♯, and (flat) A. These notes give (a mode of) the ascending melodic minor scale. Since this scale is widely used in western music, one is led to seek a relationship between the physics of overtones and the scale. Unfortunately, it turns out that the overtone corresponding to A is really closer to A♭. If a scale is constructed with the notes in the tempered scale closest to the tones in the overtone series, the results are C, D, E, F♯, G, A♭, B♭, C. This is not a scale in western music. Furthermore, the overtone associated with F♯ is almost as close to F as it is to F♯, and only tempered C, D, E, G are really close to tones in the overtone series. Thus while the overtone series may be applicable to intervals of second, third, fourth, and fifth, and to the major triad, it cannot be applicable to other intervals, the minor triad, or any scales.

8:30

DDD3. A neural network model of musical chord classification. Bernice Laden and Douglas H. Keefe (Systematic Musicology Program, School of Music, DN-10, University of Washington, Seattle, WA 98195)

A neural network, which models a musical task, specifically the chord classification task, was built. The model "listens" to a chord and classifies it as a major, minor, or diminished chord. Two basic approaches were taken in the development of the model: (1) pitch class, and (2) harmonic template. The pitch class approach represents chords as they are customarily represented in music notation, namely by specifying the pitch class of each tone in the chord. The harmonic template approach defines each pitch in terms of an equivalent harmonic complex, specified by the most appropriate pitch class label for each of the first five harmonics. This approach is motivated by pitch perception theories based upon pattern matching (Goldstein, 1973; Terhardt, 1974). Connection strengths between nodes in the network were derived using an error propagation learning algorithm (Rumelhart, Hinton, and Williams, 1986). The pitch class network performed poorly by classifying only 33%–72% of the musical chords correctly. The harmonic template network classified 100% of the musical chords correctly. [Work partially supported by the University of Washington Graduate School Research Fund and ONR Grant N00014-86-C-0065.]

8:45

DDD4. Birdsong—A quantitative acoustic model. Neville H. Fletcher (CSIRO and Research School of Physical Sciences, Australian National University, Canberra ACT 2601, Australia)

Most discussions of birdsong are at the descriptive level or, at best, involve only a qualitative discussion of the acoustical mechanisms involved in sound production. Qualitatively, two different types of song can be distinguished and these may be described as "voiced" or "whistled," respectively, though these terms do not necessarily refer to the sound-production mechanism. There is general agreement among biologists that the "voiced" song is produced by vibration of the membranes of the vocal organ (the syrinx) under the influence of aerodynamic and pressure forces, though hitherto no quantitative acoustical analysis has been attempted. The present paper puts forward such an analysis using the large body of knowledge that has been built up on the study of reed-blown musical instruments. It is shown that this approach allows a quantitative description to be formulated which yields pressure and flow waveforms and thus the form of the radiated spectrum of the song and a value for the total radiated power. Results are in quite good agreement with observations. The inability of the model to yield the nearly pure-tone sounds characteristic of "whistled" birdsong suggests that this song may be produced by an entirely different mechanism and may be truly whistled.

9:00

DDD5. Experimental developments for the measurement of violin bridge admittance. Xavier Boutillon, Gabriel Weinreich, and Nicholas R. Michael (Randall Laboratory of Physics, University of Michigan, Ann Arbor, MI 48109)

In a previous report [Weinreich and Yoder, J. Acoust. Soc. Am. Suppl. 1 **81**, S83 (1987)], a method for measuring a 2×2 admittance matrix of a violin bridge by comparing its motion with and without external loading was presented. Further investigations have shown that motion along the string direction cannot be neglected, so that a 3×3 matrix is required. Therefore, the phonograph pickup used for sensing bridge motion is replaced by three ultralight accelerometers. Their transverse sensitivity (3% to 5%) proves to have a critical effect on the validity tests for the measurements (symmetry of matrix, positiveness of dissipation). Therefore, a procedure to calibrate their sensitivity in all three directions was developed. Several possibilities have been tried as external loading: an ordinary mass, a mass attached with a wire, a spring. The first loads all three translational degrees of freedom; the others, if properly designed, only one. The approximations involved in each case will be discussed, and current results presented. [Work supported by NSF, CNRS, and French Ministry of Culture.]

9:15

DDD6. Quantitative explanation for the "flattening effect" on bowed string instruments. Xavier Boutillon (Randall Laboratory of Physics, University of Michigan, Ann Arbor, MI 48109)

McIntyre *et al.* [J. Acoust. Soc. Am. **74**, 1325–1345 (1983)] have shown through computer simulations that the "flattening effect" which occurs above a certain bow pressure level is a consequence of the hysteretical character of the stick-slip process but their time-domain analysis only indicates qualitatively why an overall increase in the period should occur. Using a frequency-domain analysis, it will be shown that the area delimited by the frictional characteristic and the string characteristic impedance is related to the spectral composition of both force and velocity at the bowing point. Because these spectral contents are also related by the string admittance at the bowing point, one can predict the flattening effect, provided that the admittance peaks are approximately harmonic. The dependence of the effect on bow velocity, termination impedances, and other parameters will be exhibited. [Work supported by NSF, CNRS, and French Ministry of Culture.]

9:30–9:45
Break

9:45

DDD7. Investigation of the acoustics of plucked string tones based on the analysis of their time-varying spectra. Kwok-ping Chen (School of Music, University of Illinois, Urbana, IL 61801)

Time-varying vibration patterns of selected tones of classical guitar, Chinese pipa, and ch'in are investigated. Based on Weinreich's theory [1977,1979] on horizontal and vertical frequencies and decay rates, partial amplitude envelopes of these tones are classified into four types. Using the normal finger-tip excitation method, it is found that certain envelope types appear more frequently in one instrument type than another. The results show that these instrument tones can be described in terms of the four types. The classical guitar open-string tone A-110 Hz is further investigated, using the pick-edge excitation method, playing at nodal position N which is L/N from the bridge. A fifth type envelope is found for the so-called missing modes [Young, 1800; Benade, 1976; Fletcher, 1984; Hall, 1987]. These modes, which are initially attenuated, subsequently rise with a more gradual attack to reach significant amplitudes.

10:00

DDD8. Biomuse: Musical performance generated by human bioelectric signals. Hugh S. Lusted and R. Benjamin Knapp (Division of Otolaryngology R-123, Stanford University School of Medicine, Stanford, CA 94305)

A new type of musical instrument is described which utilizes bioelectric signals to drive a keyboard synthesizer via standard musical instrument digital interface (MIDI) code. Electrical activity from muscle (EMG), brain (EEG), heart (EKG), and eye (EOG) is detected by small disk electrodes on the skin, analyzed by a digital signal processor, and used to control sound generation from the synthesizer. Signal processing algorithms are designed to extract useful control parameters from the given physiological signals. For instance, detection of amplitude changes in the EMG and spectral changes in the EEG can be used to generate MIDI commands. Applications for the device are discussed, such as dance generated music, musician self-mixed performances, and music production by the handicapped.

10:15

DDD9. Transposing device(s) for musical instruments. Terence S. Small (Department of Music, University of Florida, Gainesville, FL 32611)

The agonies of a concertizing musician performing upon acoustically imperfect instruments are legend. Traditional instruments (nonelectronic) continue to be manufactured largely based on the dimensions and other data derived from early times, typically through trial and error procedures. Contending with severe intonation flaws, transpositions of various sorts, and U.S. versus European pitch standards are among a performing artist's major frustrations. Efforts to achieve greater control of tone quality, pitch levels, and the like include G. Oliveri's extension core (1897), Alberti, Bender, Gemeinhardt, and a host of other inventors all striving to improve upon existing instruments. All inventions, including the writer's, deal with means by which the performer may selectively modulate the pitch and tonal quality of the instrument. In every case, the bore of the instrument is seen as the major area for manipulation, whether by altered air flow pattern or by actual real-time changes in bore dimensions. In the latter case, implications for radical instrument redesign are seen. Live musical demonstrations of a variety of these devices, with redesign commentary, form an integral portion of the presentation.

10:30

DDD10. The self-excited oscillation of and the acoustic radiation from reed instruments. P. G. Vaidya (Mechanical and Materials Engineering Department, Washington State University, Pullman, WA 99164-2920)

A harmonium reed undergoes a self-excited oscillation, when the static pressure on its two sides differs by a value, greater than a critical amount. The reed vibrates at its natural frequency, yet the sound is radiated at multiple frequencies. In this paper, a fluid mechanical model by St. Hilaire has been extended to show how the self-excitation begins, how the oscillation reaches a limit cycle, and how, due to the monopole coupling, radiation at multiple frequencies becomes possible.

FRIDAY

10:45

DDD11. Influences of the geometry of the French horn mouthpiece upon intonation. George R. Plitnik (Department of Physics, Frostburg State University, Frostburg, MD 21532) and Bruce A. Lawson (Lawson Brass Instruments, Boonsboro, MD 21713)

A mouthpiece assembly consisting of adjustable components having different dimensions, shapes, and tapers for the rim, cup, throat, and backbone was used in conjunction with a Lawson B-flat French horn for this experiment. By accurately measuring the input impedance for the mouthpiece/open French horn assembly by means of a dual-sensor apparatus [see A. M. Bruneau, J. Acoust. Soc. Am. **81**, 1168–1178 (1987)], the effect upon the amplitude and frequency of the impedance maxima could be individually gauged as each mouthpiece parameter was varied. By correlating performers' comments and subjective evaluations of the different mouthpiece combinations, an attempt has been made to relate changes in resonant frequencies and spectral envelopes to the intonation differences perceived by the player.

11:00

DDD12. An experimental comparison of the tonal qualities of a Chinese flute and a Western flute. Pan Jie (Department of Mechanical Engineering, University of Adelaide, GPO Box No. 498, Adelaide, Southern Australia 5001, Australia)

The Chinese flute has an oval, membrane-covered hole between the embouchure hole and the first tone hole. This paper presents spectrum measurements of individual notes from a Chinese flute and from a Western flute. The results indicate that the flute membrane enhances the harmonics of the flute notes and results in richer tones.

FRIDAY MORNING, 18 NOVEMBER 1988 MOLOKAI ROOM, 8:00 A.M. TO 12:07 P.M.

Session EEE. Noise III and Architectural Acoustics VIII: Active Noise Reduction

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Chairman's Introduction—8:00

Invited Papers

8:05

EEE1. An adaptive noise control system in air-conditioning ducts. Hareo Hamada, Tanetoshi Miura (Tokyo Denki University, 2-2, Kanda-Nishiki-cho, Chiyoda-ku, Tokyo, 101 Japan), Minoru Takahashi, and Yoshitaka Oguri (Research Laboratory, Hitachi Plant Engineering & Construction Co., Ltd., 537, Kami-Hongo, Matudo, 271 Japan)

This paper discusses an adaptive control system for the active cancellation of acoustic noise in air-conditioning ducts. Recently, in several systems for active noise control, efforts have been focused on the use of adaptive digital filters to implement the system controller used to produce the artificial sound. The approach used in the newly proposed model assists in canceling broadband noise and also helps in improving the adaptation speed for colored noise. This model consists of a system identification (IDT) process and an adaptive noise cancellation (ANC) process. Two stand-alone types of adaptive controllers using the fast-least-mean-squares (FLMS) algorithm and the variable-step least-mean-squares (VS-LMS) algorithm were used in various experiments. Results of experiments in actual air-conditioning ducts are presented.

8:25

EEE2. Compression system stability enhancement using active control. A. H. Epstein, E. M. Greitzer, and G. R. Guenette (Gas Turbine Laboratory, Massachusetts Institute of Technology, Cambridge, MA 02139)

All compression systems can suffer from severe dynamic instabilities, which stand as absolute limits to the system performance. The zeroth-order instability is a planar mass flow disturbance known as surge, which involves the entire pumping system [compressor or pump, ducting, plenum(s), throttle]. The high-order instabilities are called rotating stall and are local to the turbomachinery blading. Nonlinearities couple the modes together and in many practical machines rotating stall triggers surge, resulting in a large loss in performance and possible damage to the system. A large amount of effort has been expended over the last 35 years to understand these phenomena and engineer around them. More recently, work has begun on the use of active feedback control to artificially enhance compression system stability and prevent (or delay) rotating surge and stall, with particular application to aircraft engine compressors. Surge has been successfully controlled in small centrifugal compressors yielding a 20%–50% increase in operating range. Work on the stabilization of rotating stall in axial compressors is ongoing. These are very complex systems, involving hundreds or thousands of individual airfoils. Much of the work, therefore, necessarily involves signal identification in a noisy environment using sparse sampling techniques and wavelaunch with relatively few actuators. Since aircraft compressors can consume tens of megawatts, control power requirements are of concern. Analysis has shown that, in fact, the control power requirements are not tied to the size of the machine controlled but rather to the power of the destabilizing perturbations introduced to the system. Typically, the control power required is only 10^{-3} to 10^{-5} of that of the machine power and, in some cases, may be zero (the instantaneous controller power input can be either positive or negative). This paper reviews the state of the art in active compression system stabilization and is a progress report on ongoing efforts.

8:45

EEE3. Active noise control based on the inverse filtering of room acoustics. Masato Miyoshi and Yutaka Kaneda (NTT Human Interface Laboratories, 3-9-11 Midori-cho, Musashino, 180 Japan)

A novel active noise control method based on an inverse filtering theorem MINT [M. Miyoshi and Y. Kaneda, IEEE Trans. Acoust. Speech Signal Process. ASSP-36(2), 145–152 (1988)] is described. According to this theorem, broadband random noise can be precisely controlled at N points ($N = 1,2,...$) in a room using $N+1$ loudspeakers and FIR filters. Consequently, the proposed method creates a quiet zone around a given set of points. This method also has the unique property that, in a diffuse sound field, the power sum of the FIR filter coefficients is less than one. This property is employed in estimating the quiet zone attenuation. When broadband random noise is controlled at two points placed at an interval of $\lambda/4$ (where λ is the wavelength of the center frequency of the objective noise band), the estimated noise attenuation can be summarized as the following. (1) Noise attentuation of more than 14 dB is achieved at the middle of the points. (2) Over 6 dB of noise attenuation is achieved in a gourdlike quiet zone which is $\lambda/2$ long by $\lambda/8$ wide. (3) The sound-pressure level of broadband noise observed outside the quiet zone is, at most, 6 dB higher than the noise level when the noise is left uncontrolled. An experiment was conducted for controlling random noise in the range of 50–400 Hz at two points in a room in which the reverberation time was 0.5 s. The volume of the room was about 70 m^3. The experimental results showed good agreement with estimations (1), (2), and (3).

9:05

EEE4. Control criteria for active noise reduction systems. Jiri Tichy (Graduate Program in Acoustics, The Pennsylvania State University, University Park, PA 16802)

Effective control of auxiliary sources for active noise or vibration reduction can only be achieved by adaptive systems. The inputs into adaptive filters have to provide information on the signal to be canceled, as well as on the error signal from a sensor located in the area of sound field minimization. Although, in principle, some systems could achieve nearly perfect cancellation, their actual performance is limited by feedback problems from auxiliary sources which can cause system instabilities and limit the overall system performance. This problem becomes more severe with multiple feedback control which is needed, for instance, in multipath vibration transmission reduction of mechanical systems supported in several points, or multimicrophone control in enclosures. An overall analysis and approach to solutions will be presented.

9:25

EEE5. Applications for active noise control: What has been achieved and what can be achieved. Glenn E. Warnaka (Applied Acoustic Research, Calder Square, P.O. Box 10369, State College, PA 16805-0369)

Active noise control is currently undergoing an explosive growth throughout the world. This rapid development is occurring because the concept allows improvement in existing noise control devices, often with potential improvements in size, weight, volume, and cost. It also permits new solutions to noise control problems that were previously difficult or unsolvable. The technique is particularly attractive because, for many applications, noise may be reduced at the listener's position without physical modification of existing noise sources or their arrangement. Active noise control seems particularly well suited to the reduction of low-frequency noise which, for technological reasons, cannot easily or conveniently be attenuated with existing hardware. Because this new concept applies to such a wide variety of industrial, commercial, and military needs, a large number of specific applications are now under development or proposed for development. This paper describes the current status of active noise control technology. It describes applications that have already been realized as well as the possibilities for future development.

9:45–9:50

Break

9:50

EEE6. An electronic sound cancellation system for air-conditioning ducts. Hideki Hyodo, Hareo Hamada, Tanetoshi Miura (Tokyo Denki University, 2-2 Kanda-Nishiki-cho, Chiyoda-ku, Tokyo, 101 Japan), Minoru Takahashi, Ryusuke Gotohda, Yasushi Yoshimura, Taku Kuribayashi, and Akio Akasaka (Research Laboratory, Hitachi Plant Engineering & Construction Company, Ltd. 537 Kami-Hongo, Matsudo, 271 Japan)

Recently, research on an electronic sound cancellation system that is able to cancel noise using sound of the opposite phase has been conducted. This kind of noise control technique is referred to as active noise control (ANC). In order to realize a practical working ANC system, it has been considered desirable to apply the LMS (least-mean-square) algorithm to such a system's control, especially the VS (variable step)-LMS algorithm due to its high speed of convergence. The realization of an ANC system of the stand-alone type is presented. Experiments were conducted in actual air-conditioning ducts having a cross section of 500×500 mm by using the realized stand-alone type active noise control system with the VS-LMS algorithm. Below the cross-mode frequency, more than 15 dB of attenuation was obtained for broadband random noise, even in the presence of an air flow of 3.7 to 9.0 m/s.

10:02

EEE7. Application of a digital filter to active noise control in a duct. Satoshi Kuraya, Yoshitaka Nishimura, Tsuyoshi Usagawa, Masanao Ebata (Faculty of Engineering, Kumamoto University, 2-39-1 Kurokami, Kumamoto, 860 Japan), and Josuke Okda (Faculty of Engineering, Kyushu Tokai University, 223 Oh'e-Toroku, Kumamoto, 862 Japan)

The possibility of active noise control and its frequency range are discussed using sound-source impedance. When a digital filter is used in an active noise control system, the delay time required in the filter must be longer than those in AD-DA converters and low-pass filters. In order to make a two-microphone active control system possible and to lower its frequency range, it is required that (1) the distance between the microphone set near the original source and the loudspeaker used for an additional source should be far; and that (2) the sampling frequency should be high. However, as the number of digital filter taps is fixed, there is a limitation on the sampling frequency in order to realize the characteristics in the low-frequency range. In the proposed system with motional feedback, the active control is realized in a frequency range lower than the fundamental resonance frequency of the loudspeaker, f_0. Thus f_0 must be raised in order to realize a system at a higher frequency range.

10:14

EEE8. Adaptive digital filter configurations for active control of lightly damped systems. D. E. Waters and R. J. Bernhard (School of Mechanical Engineering, Purdue University, West Lafayette, IN 47907)

The application of system identification techniques using adaptive digital filters to achieve active noise control in one-dimensional duct systems has been previously investigated [J. C. Burgess, J. Acoust. Soc. Am. **70**, 715–726 (1981), L. J. Eriksson, Ph.D. thesis, University of Wisconsin, Madison (1985)]. In this investigation, efficient system identification techniques using adaptive digital filters for active control of lightly damped systems at low modal density are sought. Three cases are considered: (1) where feedback to the detector sensor exists but where the error path is negligible, (2) where the error path exists but feedback is negligible, and (3) where both feedback and error paths are important. In each case, the active controller uses two system identification algorithms in a manner similar to that developed by Eriksson. The first algorithm models the error path by driving the secondary source with its own random noise generator and sensing the resulting response at the error transducer. The error path model is shared with the second system identification algorithm which uses a detection transducer signal, the error signal, and the estimated error path to model the plant (and feedback, if necessary). In each case, both FIR and IIR filters are considered. [Work supported by Nelson Industries, Stoughton, WI.]

10:26

EEE9. A study of active control of sound transmission through a panel into a cavity. Pan Jie (Department of Mechanical Engineering, University of Adelaide, GPO Box No. 498, Adelaide, Southern Australia 5001, Australia)

A technique for controlling noise transmitted into the interior of a cavity involves use of point force actuators on the boundary structures. This paper is a study of the optimal actuator positions and the maximum theoretical reduction of the sound transmission through a panel into a rectangular cavity. Results obtained demonstrate that the noise reduction by the actuators depends upon the nature of the incident sound. Details of the manner in which the control forces act on panel–cavity system are also discussed.

10:38

EEE10. An active noise control system by means of motional feedback without a microphone. Yoshitaka Nishimura, Tsuyoshi Usagawa, Masanao Ebata (Faculty of Engineering, Kumamoto University, 2-39-1 Kurokami, Kumamoto, 860 Japan), and Josuke Okda (Faculty of Engineering, Kyushu Tokai University, 223 Oh'e-Toroku, Kumamoto, 862 Japan)

A method for active noise control using a motional feedback system without any microphone is described. The impedance of an electroacoustic transducer can be controlled by motional feedback, and the noise in a duct can be reduced actively by adjusting the impedance of the transducer used for an additional sound source. The characteristics of the impedance of the transducer with the motional feedback and noise reducing effect of this system are analyzed, and they are measured using both analog filter implementation and a digital one. The results show that this system is effective in controlling the noise in a duct, but it is not applicable to a wide frequency range. The advantage of this system is that it does not need any microphone which has been a weak element in conventional active control systems under undesirable circumstances, such as high pressure, high temperature, dust, air flow, and so on.

10:50–10:55

Break

10:55

EEE11. An active noise canceling microphone. Glenn E. Warnaka (Applied Acoustic Research, Calder Square, P.O. Box 10369, State College, PA 16805-0369)

This paper describes a noise canceling microphone that uses the principles of active noise control. Both physical (acoustic) cancellation, by means of a loudspeaker, and electronic cancellation have been applied to the problem. The results show that a high level of performance may be obtained while removing the requirement to have the microphone extremely close to the lips. Excellent cancellation was achieved at a distance of 24 in. from the signal source using tones, harmonic noise, pink noise, and a variety of cockpit noises from a helicopter, a turbo-prop, and a jet fighter. Conventional microphones may be used to implement the active noise canceling microphone, or a conventional noise canceling microphone may also be used as a basis for implementation of this concept. The active noise control microphone is superior to conventional noise-canceling microphones in rejecting background noise.

11:07

EEE12. Active noise control on systems with time-varying sources and acoustical elements. L. J. Eriksson, M. C. Allie, C. D. Bremigan, and J. A. Gilbert (Nelson Industries, Inc., P.O. Box 600, Stoughton, WI 53589-0600)

An active noise control system has been developed based on a system identification concept using a recursive adaptive filter to model the direct and feedback elements of an acoustic system. An independent random noise source is used with a second adaptive filter to provide a model of the output transducer and error path transfer functions. This system has the ability to continuously cancel narrow band and broadband noise. The system identification configuration combined with adaptive filter techniques enables the system to respond quickly and accurately to changes in time-varying sources, such as frequency or amplitude, and changes in the acoustical system, such as temperature. Noise reduction results will be presented. The system capabilities will be demonstrated on a variety of time-varying sources and time-varying acoustical elements. The ability of the system to track these changes in real-time as well as implications for actual applications will be discussed.

11:19

EEE13. Consideration of active noise control in space. Josuke Okda (Faculty of Engineering, Kyushu Tokai University, 223 Oh'e-Toroku, Kumamoto, 862 Japan), Tsuyoshi Usagawa, and Masanao Ebata (Faculty of Engineering, Kumamoto University, 2-39-1 Kurokami, Kumamoto, 860 Japan)

Active noise control in space is classified into four groups based on physical aspects, which can be further reduced into two groups, namely, the sound pressure or the sound power, and the ability or nonability of an additional sound source to act on the sound radiation of an original source. The latter is equivalent to the problem of whether the mutual impedance between the two sound sources must be considered or not. The noise reduction process in each group is analyzed using the impedance of the sources. The desirable condition of the additional source is discussed, and, when this condition is realized, the noise reduction effect is estimated. In general, when the additional source is placed near the original one, the more tightly they are coupled, the more effectively both the sound pressure and the sound power are reduced. It is difficult, however, to find the most desirable condition for reducing the sound power by means of an experimental procedure.

11:31

EEE14. Active noise absorption as an inverse source problem. Woon S. Gan (Acoustical Services Pte Ltd., 29 Telok Ayer Street, Singapore 0104, Republic of Singapore)

In an ordinary approach to three-dimensional active noise absorption (ANA), the degree of noise cancellation will increase with the increasing number of small secondary sources used. Here, a three-dimensional ANA is formulated as an inverse source problem and acoustical holography is used to realize the secondary (absorbing) sources. The advantage is that one does not have to approximate the number of secondary sources used to obtain identical field to the primary source which is a prerequisite for ANA. The use of generalized holography is proposed. Here, the generalized hologram is constructed by recording the field and its normal derivative over a closed surface surrounding the primary noise source. This gives the Huygens' sources. The required secondary sources are obtained by the reconstruction of the generalized hologram, by allowing the recorded field and normal derivative to backpropagate into the space region containing the primary source. To test the theory, computer simulation is proposed. The three-dimensional primary noise source chosen is a transformer noise source. The required simulation algorithm is derived in terms of a deconvolution process.

11:43

EEE15. The use of electronic noise cancellation in the mining industry. Leonard C. Marraccini and Dennis A. Giardino (Mine Safety and Health Administration, 4800 Forbes Avenue, Pittsburgh, PA 15213)

In order to reduce the noise exposure of people working in the mining industry, various types of noise control applications have been used. These have included standard retrofit noise controls, machinery redesign, and administrative controls such as modifying employee work cycles. Currently, the Mine Safety and Health Administration is investigating the use of electronic noise cancellation techniques for reducing worker noise exposure. This paper characterizes this work including background information, a description of the laboratory development work involving electronic noise cancellation, and the application of this technology in a field situation. By using the electronic cancellation system and loudspeakers, a zone or area of noise cancellation is produced.

11:55

EEE16. Mechanisms of active sound control. Colin H. Hansen, Scott D. Snyder, and David A. Bies (Department of Mechanical Engineering, University of Adelaide, G.P.O. Box 498, Adelaide, South Australia 5001, Australia)

Active noise control applied to sound propagating down ducts has been successfully demonstrated by a number of researchers. The active control of small three-dimensional sound fields (such as those found in ear muffs) has also met with considerable success. However, application of active control techniques to large three-dimensional sound fields, such as those found in rooms and vehicle interiors, has only been partially successful, generally resulting in sound reduction in some locations at the expense of an increase in other locations. The lack of success is easily explained if the mechanism by which successful active control is achieved is fully understood. This paper presents the results of an experiment that demonstrates the mechanism by which successful active control is achieved in a duct and conclusions are made regarding the requirements for successful active control in large three-dimensional spaces, which would enable an overall, rather than just a local, reduction in sound level to be achieved. [Work supported by the Sir Ross and Sir Keith Smith fund and the University of Adelaide.]

Session FFF. Physical Acoustics VII: Scattering

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Contributed Papers

8:00

FFF1. Nonspecular reflection of sound by elastic bodies and wave front reversal. Leonid M. Ljamshev (N. N. Andreev Acoustical Institute of the Academy of Sciences of the USSR, Moscow, USSR)

Results of experimental and theoretical research of nonspecular reflection of sound (NRS) by bounded thin or thick plates, rods, and shells in water are discussed. The results had been completed and published by the author during the 1950s [L. M. Ljamshev, Akust. J. **2**, 189, 228 (1956); **5**, 58 (1959); Dokl. Akad. Nauk **99**, 719 (1954); **110**, 48 (1956) and so on, see also W. Finney, J. Acoust. Soc. Am. **29**, 625 (1948)]. These results are compared with the results of NRS of ultrasonic-bounded beams by thick plates in liquids. This effect was observed and described later during the 1970s and 1980s [W. Neubauer, J. Appl. Phys. **44**, 48 (1973); M. Billy and L. Adler, J. Appl. Phys. **75**, 393 (1984) and in the book: *La Diffusion Acustique* (CEDOCAR, Paris, 1987)]. It is shown that these effects of NRS are of an identical physical (linear) nature. A new type of NRS is discussed. It arises as the result of nonlinear interaction of an incident sound wave with a vibrating body when vibration frequency equals twice the frequency of incident sound wave and the phenomena of wave front reversal appears.

8:12

FFF2. Acoustic scattering by a sound-hard rectangle. Anders Boström (Division of Mechanics, Chalmers University of Technology, S-412 96 Göteborg, Sweden)

The scattering of an incoming plane wave by a sound-hard infinitely thin rectangle is considered. An integral equation method which has previously been used for two-dimensional problems [A. Boström, J. Appl. Mech. **54**, 503–508 (1987)] and three-dimensional axisymmetric problems [S. Krenk and H. Schmidt, Philos. Trans. R. Soc. London Ser. A **308**, 167–198 (1982)] is employed. Starting from a double spatial Fourier field representation, a matching of the conditions in the plane of the rectangle leads to an integral equation for the potential jump across the rectangle. This jump is expanded in a double series of Chebyshev polynomials which fulfill the right edge conditions (but no special measures are taken for the corners where the right conditions are unknown anyway). The integral equation is thus discretized and the problem thereby solved. By a double stationary-phase analysis the farfield is determined and some numerical examples of this are given.

8:24

FFF3. Equivalent-network representation of an infinite elastic plate subject to plane-wave excitation. Anthony J. Rudgers (Naval Research Laboratory, Underwater Sound Reference Detachment, P.O. Box 568337, Orlando, FL 32856-8337)

A plane wave incident upon an elastic plate at an oblique angle excites both dilatational and shear waves in the plate. On reflection at the plate surfaces, shear waves with displacement components nonparallel to the plate surfaces are partially converted to dilatational waves, and dilatational waves are partially converted to such shear waves. The mechanical behavior of a plate in such circumstances can be modeled, not only by means of equations derived using the theory of linear elasticity, but also with a four-port network that is equivalent to these equations. This network comprises two coupled transmission lines, one representing the dilatational and the other representing the shear-wave propagation. A pair of ports at each plate surface transmits the normal and the transverse forces and displacements appearing at that surface. Each port is connected to the two transmission lines through a network of ideal transformers. A layered elastic media can be represented by a number of such equivalent networks connected in cascade. [Work supported by ONR.]

8:36

FFF4. The k-space formulation of acoustic scattering using the conjugate-gradient method. Shozo Koshigoe (Code 3892, Naval Weapons Center, China Lake, CA 93555), Norbert N. Bojarski (1320 Santiago Drive, Newport Beach, CA 92660), and Arnold Tubis (Department of Physics, Purdue University, West Lafayette, IN 47907)

An extension of the iterative k-space method [N. Bojarski, J. Acoust. Soc. Am. **72**, 570–584 (1982)] is used to calculate acoustic scattering. The conjugate gradient method, with guaranteed convergence, is used instead of the Neumann series for the iterative calculations. In order to accelerate the computation and reduce the memory-size requirement, the convolution type integral involved in the use of the conjugate gradient method is handled as in the k-space method. One-dimensional acoustic scattering by an arbitrary slab is calculated as a simple illustrative application of the formalism. In order to demonstrate the utility of the method, some of the calculated results are compared with those obtained by the Neumann method.

8:48

FFF5. Compressional and shear wave fields in an elastic sphere. Michael J. Buckingham (Mission Management Department, Royal Aerospace Establishment, Farnborough, Hampshire GU14 6TD, England)

An exact solution for the longitudinal and transverse wave fields generated by a point monopole source in a solid, elastic sphere *in vacuo* has been developed. The solution for the scalar and vector potentials is valid throughout the sphere, including the source point. The simple pattern of resonances that would occur in the compressional wave in the absence of shear, vanishes when elasticity is introduced. It is replaced by a much more complicated resonance structure. As a result of coupling at the spherical boundary, the new resonances appear at the same values of the Helmholtz number in the compressional and shear waves. Both fields show a highly structured spatial dependence, in which a predominant feature is radial banding: on tracking out from the center of the sphere to the periphery, the amplitude of the fields alternates in sign. The analytical technique developed for this problem has also been applied to the case of a fluid loaded sphere. This is relevant to sound generation by ice floes in the Arctic Ocean, and to controlled experimental studies of acoustic emission from spherical specimens.

9:00

FFF6. Product expansion of the S function for scattering from elastic spheres having multiple resonances. Philip L. Marston (Department of Physics, Washington State University, Pullman, WA 99164)

Associated with the scattering phase shift δ_n of the nth partial wave for a sphere of radius a is the function $S_n(x) = \exp[2i\delta_n]$, where $x = ka$. A theorem from *classical* scattering theory [H. M. Nussenzveig, *Causality and Dispersion Relations* (Academic, New York, 1972), pp. 54–72; N. G. van Kampen, Phys. Rev. **89**, 1072–1079 (1953)] leads to the following *product expansion* which appears to be important and novel for acoustics:

$$S_n = \pm e^{-2ix} \prod_l \frac{(x_{nl}^* - x)(x_{nl} + x)}{(x_{nl}^* + x)(x_{nl} - x)} \prod_j \frac{(iL_{nj} - x)}{(iL_{nj} + x)},$$

where the $x_{nl} \equiv X_{nl} - i(\Gamma_{nl}/2)$ lie in the fourth quadrant and $L_{nj} > 0$. Unlike a *sum* expansion (in some acoustic RST literature) S_n remains manifestly unitary even for multiple resonances l. This S_n may be used to split off elastic contributions to the partial-wave form function f_n for backscattering in analogy with RST. In the case of only two resonances (labeled $l = 1$ and 2) $f_n \approx f_n^{(b)} + f_{n1} + f_{n2} + f_n^{(\mathrm{int})}$, where $f_n^{(b)}$ is a background term and it is assumed that $x + X_{nl} \gg \Gamma_{nl}$. The f_{nl} have a Breit–Wigner form $f_{nl} = \exp(2i\xi_n)[(2n+1)/x](-1)^n \Gamma_{nl}/[X_{nl} - x - (i/2)\Gamma_{nl}]$ and ξ_n is the phase shift associated with $f_n^{(b)}$. The *interaction term* $f_n^{(\mathrm{int})}$ vanishes as $\Gamma_{nl}/|x - X_{nl}| \to 0$ for $l = 1$ or 2. This clarifies implicit assumptions of formal RST. [Work supported by ONR.]

9:12

FFF7. Graphical representations of the spherical-shell-form-function based on Wigner distribution algorithm. N. Yen, Louis R. Dragonette, and Susan K. Numrich (Physical Acoustics Branch, Naval Research Laboratory, Washington, DC 20375-5000)

The reflection of acoustic waves from a spherical shell is a well-defined scattering problem that can be formulated mathematically in terms of a normalized response either in the frequency or in the time domain. Studies made from computer generated graphic displays, based on Wigner distribution algorithm, provide a clear and detailed representation of the underlying physical processes which might otherwise be incomprehensible because of the complexity of the high-order mathematical functions. The sensitivity of such graphical representations with regard to the sampling size and frequency bandwidth is analyzed with some examples.

9:24

FFF8. An efficient and accurate numerical technique for low-frequency scattering from elastic bodies. Harry A. Schenck and George W. Benthien (Ocean Surveillance Department, Naval Ocean Systems Center, San Diego, CA 92152-5000)

In the low-frequency regime where the dimensions of the object and the acoustic wavelength are comparable, it is possible to model the bistatic scattering from an elastic body numerically with high spatial and spectral resolution. This paper will describe a computer model that is built around a finite-element program (MARTSAM) which provides an accurate description of a general elastic body and a boundary element or integral equation method (CHIEF) by which the acoustic scattering from a body of arbitrary shape can be calculated. An additional program has been developed and implemented to couple the CHIEF and MARTSAM outputs to provide accurate solutions of the combined structural acoustic scattering problem for objects of arbitrary shape with internal elastic structure. These programs have been implemented on a range of computers including supercomputers. Examples will be given that illustrate the accuracy and resolution of the method and of the computing time as a function of frequency.

9:36

FFF9. Bistatic scattering from elliptical shells. Ronald P. Radlinski (Naval Underwater Systems Center, New London, CT 06320) and Murray Simon (Honeywell, Inc., Seattle, WA 98107)

The extended boundary integral method developed for elastic wave scattering from infinite cylindrical shells of arbitrary shape [M. M. Simon and R. P. Radlinski, J. Acoust. Soc. Am. **71**, 273–281 (1983)] was modified to investigate bistatic scattering from thin elliptically cylindrical shells in fluid. Thin shell theory is assumed to describe the motion of the scatterer. Expansion of the normal and tangential displacements of the shells in Fourier series which are a function of arclength reduces the shell equations to algebraic form. Scattering cross sections and bistatic scattering are shown to be a function of the orientation and material of the shell. Extensional resonances are shown to have a wide bandwidth influence on the scattering patterns, but narrow-band resonance is also found with certain materials.

9:48

FFF10. The acoustic scattering by a submerged, elastic spherical shell—Pole trajectories in the complex k plane. Roger H. Hackman and Gary S. Sammelmann (Physical Acoustics Branch, Naval Coastal Systems Center, Panama City, FL 32407)

At a previous meeting of the society [G. S. Sammelmann and R. H. Hackman, J. Acoust. Soc. Am. Suppl. 1 **83**, S94 (1988)], a fundamentally oriented analysis of the pole structure of the acoustic scattering amplitude for an elastic spherical shell was presented. In this presentation, it was demonstrated that fluid loading the first antisymmetric mode of a spherical shell in vacuum had a rather violent effect; the dispersion curve bifurcates near the frequency that the vacuum curve transitions from a subsonic to a supersonic phase velocity. Here, the precise relationship of the Franz wave poles of rigid scattering and the poles corresponding to the vacuum eigenvibrations to the acoustic scattering poles of the fluid-loaded spherical shell varying the elastic sound speeds of the shell are established. The fluid-loaded modes are found to "switch tails" under variation of the sound speeds and it is found that the unique identification of a given mode as either "diffractive" or "elastic" in character is not generally possible.

10:00

FFF11. Low-frequency elastic excitations of large aspect ratio, solid elastic targets and their couplings to the acoustic field. Roger H. Hackman, Gary S. Sammelmann, and Kevin L. Williams (Physical Acoustics Branch, Naval Coastal Systems Center, Panama City, FL 32407)

There is currently a controversy concerning the nature of the elastic resonances of large aspect ratio solids and the couplings of these resonances to the acoustic field. Previously [Williams *et al.*, J. Acoust. Soc. Am. Suppl. 1 **82**, S50 (1987)], both theoretical and experimental evidence was presented to support our contention that the elastic wave underlying the low-frequency resonances is essentially a "bar wave" with a phase velocity almost twice that of the shear speed. At this meeting, further evidence to this effect is presented and is carefully examined by the couplings of the elastic and acoustic fields. It is concluded that the excitation of the low-frequency axisymmetric resonances of these targets is not due to phase matching, but that the coupling can be best described in terms of "impulse excitations." Also established is the relationship of the prominent, off-axis radiation lobes of these resonances to the surface displacements of the bar wave and it will be demonstrated that these lobes are not consistent with the Rayleigh surface wave interpretation.

10:12

FFF12. An analysis of the off-axis scattering from solid prolate spheroids. Roger H. Hackman and Gary S. Sammelmann (Physical Acoustics Branch, Naval Coastal Systems Center, Panama City, FL 32407)

The acoustic scattering from solid, elastic prolate spheroids is studied as a function of frequency, aspect angle, and aspect ratio. The emphasis of this study is on the prolate spheroid as a transition geometry. The interfocal distance is varied from zero (the spherical limit) to almost unity, as the semimajor axis is held fixed. In this way, the smooth transition from spherical phenomena to what is essentially an infinite cylindrical limit is studied, and the relation between the spherical and infinite cylindrical elastic waves is established. It is argued that the more robust features of the solution for a large aspect ratio spheroid are common to all slender, elongated, axisymmetric solid scatterers of the same aspect ratio, particularly at low frequencies (i.e., low $kL/2$). It is in this latter sense that the prolate spheroidal solid is expected to be a "generic" large aspect ratio, solid scatterer.

10:24

FFF13. Lamb waves and fluid-borne waves on water-loaded thin spherical shells. Russel D. Miller (NKF Engineering, Inc., 12200 Sunrise Valley Drive, Reston, VA 22091), Maryline Talmant, Jaime Castillo, Herbert Überall (Physics Department, Catholic University of America, Washington, DC 20064), and M. F. Werby (Naval Ocean R&D Activity, Code 221, NSTL Station, MS 39529-5004)

A recent study of fluid-borne waves on plates with one-sided fluid loading [M. Talmant, Ph.D thesis, University of Paris VII (1987)] allows the prediction of the corresponding waves and their resonances (as well as of the Lamb-wave resonances) on thin submerged spherical shells. Similar fluid waves and the ensuing bifurcation in the dispersion curves of the first antisymmetric vibration mode (reminiscent of the repulsion of atomic levels) on cylindrical shells were previously described [Breitenbach *et al.*, J. Acoust. Soc. Am. 74, 1267 (1983); J. V. Subrahmanyam, Ph.D. thesis, Catholic University (1983)] and observed [Talmant *et al.*, J. Acoust. Soc. Am. (in press)]. The predicted resonances were confirmed by scattering cross section calculations using SIERRAS (R.D.M.) and *T*-matrix (M.F.W.) codes.

10:36

FFF14. Resonant eigenfrequencies of elastic spheroids and the "level crossing" phenomenon. M. F. Werby (Naval Ocean R&D Activity, Code 221, NSTL Station, MS 39529-5004), J. Castillo (Physics Department, Catholic University of America, Washington, DC 20064), Russel Miller (NKF Engineering Inc., 12200 Sunrise Valley Drive, Reston, VA 22091), J. W. Dickey (David W. Taylor Research Center, Annapolis, MD 21402), and H. Überall (David W. Taylor Research Center, Annapolis, MD 21402 and Physics Department, Catholic University of America, Washington, DC 20064)

The "level crossing" phenomenon, which has been pointed out earlier, is studied here for solid elastic spheroids of varying aspect ratios. Using the NORDA *T*-matrix code, the frequencies (both fundamentals and overtones) of the resonances are calculated as excited by an end-on incident plane wave. Aspect ratios of the tungsten carbide spheroids range from $b/a = 1$ (sphere) to 4 in steps of 0.25. The resonance frequencies are expressed in units kb (b = semimajor axis) and correspond to resonating surface waves as they form standing waves around the meridional circumference. While for the resonating Rayleigh waves, the resonance frequencies only rise gradually with increasing aspect ratio, those for the Whispering Gallery waves rise much more rapidly, and, hence, successively cross over the various overtones of the Rayleigh resonances. This phenomenon is explained on the basis of the different character of the dispersion curves (phase velocity versus frequency) for the Rayleigh and the Whispering Gallery waves, as calculated in an exact fashion for a WC sphere and used locally on the spheroids.

10:48

FFF15. Solving the inverse scattering problem to obtain acoustic bubble spectra. Elan Moritz (NCSC Code 2230, Panama City, FL 32407-5000) and Kerry W. Commander (NCSC Code 2120, Panama City, FL 32407-5000)

Determination of acoustic bubble spectra near the surface of the ocean, using resonance scattering approximations, leads to considerable overestimation of the numbers of bubbles with radii less than 50 microns [J. Acoust. Soc. Am. Suppl. 1 82, S109 (1988)]. Several numerical methods for solving the exact inverse scattering problem were examined and results for each are presented. In these studies, attenuation and backscattering strengths, as a function of frequency, were used as input in each of the techniques, and the computer bubble distribution was compared with known (*a priori*) distributions. Sensitivity of the numerical techniques to dependence on the attenuation and backscattering functions was explored for each method as well. Improvements over resonance theory approximations are shown through a systematic study of each technique using known bubble distributions. [Work supported by ONT.]

11:00

FFF16. Inverse problem and scattering from elastic cylinders at high frequencies. G. Quentin, F. Luppé, and A. Cand (G.P.S., University Paris 7, Paris, France)

For values of $k_1 a$ larger than 200, it has been previously shown that the echoes backscattered from a cylinder can be explained using a geometrical theory [Quentin *et al.*, J. Acoust. Soc. Am. 70, 870–878 (1981)]. Now the problem of the identification of an elastic cylindrical scatterer from the set of backscattered echoes is addressed. It is shown that from a systematic research, it is possible to deduce the ratio C_L/C_T of the longitudinal to the transverse velocity in the solid with a very good precision and with less precision for these velocities, the density of the material, and the radius of the cylinder. The advantages of applying to this problem methods deduced from artificial intelligence is also discussed.

11:12

FFF17. Acoustic waveforms in a fluid-filled borehole with axially periodic thin rigid cylinders of finite length. S. K. Chang (Schlumberger-Doll Research, Old Quarry Road, Ridgefield, CT 06877-4108)

The interference of acoustic waveforms in a fluid-filled borehole by a periodic structure in the hole is investigated theoretically. The structure consists of a series of finite length thin rigid hollow cylinders periodically located along and concentric with the axis of the hole. Numerical results are obtained using the Galerkin's method. The radial displacement field in the gaps between the sections of the rigid cylinders is expanded in a set of basis functions that satisfy the singularity condition at the edges of the cylinders. The time domain signals are calculated for a point source and point receivers distributed on the hole axis inside the periodic structure. The interference effects of the structure are observed by varying the period and the height of the hollow cylinderical sections. It is found that the interference is dominated by the open-cavity resonance of the hollow cylinders. The resonance can be avoided by choosing proper dimensions of the cylindrical sections. Without the cavity resonance, the interference is minimized and the periodic structure is almost acoustically transparent in the borehole.

11:24

FFF18. A laboratory model for scattering by a finite thin plate. Chaur-Jian Hsu and S. K. Chang (Schlumberger-Doll Research, Ridgefield, CT 06877-4108)

Time domain acoustic wave scattering by a thin metal plate with a finite width is studied experimentally in a water tank. The plate is illuminated by a point pressure source, and the scattered waves are detected by a hydrophone. The source and receivers are located along a line which has a projection on the plane of the plate perpendicular to the edges of the plate. Experiments are conducted at several angles between the plate and the source–receiver line. The waveforms in the space-time domain clearly show three main events: a flat surface reflection from the plate, and two edge diffractions. The reflected waves show up as a straight line segment indicating the angle between the source–receiver line and the plane of the plate. The diffracted waves in the space-time domain have curvatures that are consistent with the location of the edge diffraction predicted by ray theory. The multiple ringings of the plate scattering are also visible as secondary events in the later time.

11:36

FFF19. A characterization of the power spectrum of scattered ultrasound using a fractal-based analysis. Tsuneo Kikuchi, Shogo Kiryu, Sojun Sato, and Hajime Miura (Electrotechnical Laboratory, 1-1-4 Umezono Tsukuba, 305 Japan)

A method for extracting the characteristics of the power spectrum of ultrasound scattered from multiple scatterers using a fractal-based analysis is presented. The shape of the power spectrum is usually very complicated. The shape contains structural information about the medium, because the shape depends on the internal structure of the medium. If the power spectrum of the scattered ultrasound has a fractal structure, characteristics such as self-similarity and fractal dimensions are useful to characterize the power spectrum. When the scatterers are regularly arrayed, a high self-similarity of the power spectrum will be expected. On the other hand, when the array is partially disordered, the self-similarity of the power spectrum will be lower. Fractal dimensions will be expected to relate to the mean distance between the scatterers. By numerical simulations and basic experiments, the validity of these ideas has been confirmed.

11:48

FFF20. Piecewise constant optimal controls and minimal acoustic transmission. Kurt P. Scharnhorst (Naval Surface Warfare Center, White Oak, Silver Spring, MD 20903-5000)

The existence of completely piecewise constant optimal controls in the wide bandwidth, normal incidence, minimal acoustic transmission problem [MIN(γ^2)] for layers of viscoelastic materials is examined numerically. It has been found that such solutions exist if the Hamiltonian is linear in the controls and the dynamic compressibility is defined in terms of fractional derivatives which approximate a generalized Maxwell model of the dilatation modulus. The low-frequency dynamic compressibility and the low- to high-frequency step of the compressibility in the relaxation region are assumed to be the controls. Calculations are presented that show the resulting optimal, multilayered structures. For shallow slopes of the compressibility the optimal controls are partially continuous (singular). As the slope increases, the controls become piecewise constant. Piecewise constant optimal controls are the exception. Their appearance seems to be restricted to relatively simple definitions of the control functions. Their occurrence may also be limited to the MIN(γ^2) problem in the set of three basic problems defined by transmission, reflection, and absorption.

12:00

FFF21. Acoustic radiation force on a rigid sphere in the nearfield of a circular piston vibrator. Takahi Hasegawa, Naoki Inoue, and Kiichiro Matsuzawa (Department of Physics, Faculty of Science, Ehime University, Matsuyama, 790 Japan)

The acoustic radiation force on a rigid sphere in the nearfield of a circular piston vibrator was investigated theoretically. The model considered is a rigid sphere suspended freely in an inviscid fluid, the center of the sphere being on the axis of a circular piston vibrator excited harmonically in an infinite baffle. The velocity potential in the nearfield was represented as an infinite series of the spherical harmonics by use of the diffraction integral for the vibrator. The result showed that the acoustic radiation force is given in the form of an infinite series as a function of the scattering coefficient and the diffraction integral. The nth diffraction integral fn is given in the form of a recursion formula as a function of the parameters of the vibrator and the distance between the sphere center and the vibrator. The formula obtained is valid for any elastic solid spheres. The present study will have useful applications in absolute measurements of ultrasound intensity in the nearfield at megahertz frequencies.

Session GGG. Speech Communication XII: Special Focus Session III, Speech Processing in Human–Machine Interaction—An International View

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Chairman's Introduction—8:00

Spoken language is no doubt the most natural and efficient means of human communication, and it will therefore be the ultimate medium for the efficient exchange of a large amount of information between man and machine. With the increase in the understanding of the basic properties of speech and natural language on the one hand, and the rapid progress in techniques for signal and information processing on the other, man–machine communication through spoken language has become a realistic goal for applied research. Since, however, it requires an integration of research efforts in a number of related fields, several large-scale projects have been initiated recently to combine and coordinate research activities either at the national level or at the international level, often in connection with research towards future generation computer technology. This session will provide both an overview of these large-scale projects by their representatives, and a forum for discussion on future prospects, as well as on exchanges and cooperation.

Invited Papers

8:05

GGG1. Overview of Japanese efforts toward an advanced man–machine interface through spoken language. Hiroya Fujisaki (Department of Electronic Engineering, Faculty of Engineering, University of Tokyo, 7-3-1 Hongo, Bunkyo-ku, Tokyo, 113 Japan)

An overview will be given of some recent efforts in Japan to promote and coordinate research activities toward the realization of an advanced man–machine interface through spoken language. In the field of speech processing, a large-scale national project was started in 1987 by a Grant-in-Aid for Scientific Research on Priority Areas from the Ministry of Education, Science and Culture (MESC) of Japan. It involves all the important academic groups as well as a number of experts from governmental organizations and private industries, and emphasizes the systematic coordination of research efforts through the use of common speech data, standardization of research tools, and shared research results. Another national project will start in 1989 to construct a database of the spoken dialects of Japanese. In the field of natural language processing, a separate project has also been carried out with the support of MESC. Efforts toward an integration of studies in the two areas, as well as toward closer cooperation between academic and industrial circles, have resulted in the establishment of a committee for academia–industry cooperation on the intelligent processing of linguistic information both in spoken and written forms under the auspices of the Japan Society for the Promotion of Science. Some highlights of these activities will be given in the lecture.

8:25

GGG2. The DARPA spoken language systems program: Past, present, and future. J. Allen Sears (Intelligent Systems Program, DARPA-ISTO, 1400 Wilson Boulevard, Arlington, VA 22209)

The long-range goal of the DARPA strategic computing speech program is to develop speech recognition and understanding technology sufficiently robust to support a wide range of DoD applications. The application environments may require systems capable of handling up to 10 000 words, with habitable grammatical constraints measurable by a perplexity of 100 to 200. Such a system should have a word accuracy of better than 95%, and should operate in real time. Specific short-term goals for the first phase include the development of a 1000-word continuous speech recognition system for use in acoustically benign environments by the end of 1988. Research activities supported by the DARPA strategic computing speech program include the development of speech recognition systems, related component technology, and supporting infrastructure. For the next 4 years, the program will build on accomplishments of phase one of the speech recognition and natural language programs in order to achieve spoken language understanding. Specifically, the program will move from speech transcription and textual language processing towards interactive problem solving using voice input. In this talk, the objectives and the structure of the DARPA-sponsored program will be described in greater detail.

8:40

GGG3. Technical achievements of the DARPA speech recognition program. Victor W. Zue (Room 36-575, Department of Electrical Engineering and Computer Science, Massachusetts Institute of Technology, Cambridge, MA 02139)

During the first 4 years, several significant milestones have been reached by participants of the DARPA strategic computing speech program in the development of recognition systems, related component technology, and common speech databases and research tools. From the systems' standpoint, speaker-dependent, -adaptive, and -independent systems that operate in several times real time with word error rates that range between 5% and 10% have been demonstrated with a 1000-word vocabulary and a language model perplexity of 60. Real-time performance of isolated-word, speaker-dependent, 100-word recognition systems that operate under noisy and stressful environments was demonstrated with word accuracy of better than 98%. Meanwhile, advances have been made in component technology including auditory modeling, acoustic phonetic recognition, lexical representation, language modeling, and the implementation of algorithms in general- and special-purpose hardware. The component technology is intended to provide robust performance in real time, and to support future expansion in task complexity. The program also resulted in the development of several large databases, and a set of common research tools. The use of these databases in conjunction with the development and implementation of standardized performance test procedures has permitted consistent documentation of progress. This talk will describe these technical areas in greater detail. [Work supported by DARPA-ISTO.]

8:55

GGG4. Speech processing in human–machine interaction: Research in the United Kingdom. Frank Fallside (Cambridge University Engineering Department, Cambridge CB2 1PZ, United Kingdom)

Research into speech processing in human–machine interaction in the U.K. has been much stimulated by the Alvey initiative in advanced information processing. This will shortly be succeeded by the Information Engineering Directorate (IED) of the Department of Trade and Industry with support from other government bodies, including the Science and Engineering Research Council, together with support from the second phase of the European Economic Community's ESPRIT initiative. The paper describes the broad areas of research of these and other initiatives in the U.K. under three headings: (a) enabling research into speech recognition and synthesis, (b) systems, and (c) infrastructure. Work on speech recognition using the techniques of hidden Markov models, feature based methods, hybrid classifier, and connectionist models/neural networks is described, together with research into text to speech synthesis in several laboratories. The Alvey MAST system for speech transcription and VODIS system for database inquiry are described. A brief description is given of the work on speech assessment under the STAG and SAM projects. Finally, the IED strategy with its emphasis on multidisciplinary research is described.

9:15

GGG5. Program of the European communities for the European speech technology. J. Roukens (Commission of the European Communities, Rue Archimède 25-3/10, B 1049 Brussels, Belgium)

Considering the strategic importance and potential size of the speech technology market, the European Commission helps the technological advance of laboratories and industries in this area by financing a series of projects within the ESPRIT program. These projects bring the best European laboratories and advanced industries together for cooperation in this area, and are oriented in four directions: (1) acquisition of basic knowledge about languages and human factors in human–machine speech dialogue; (2) advancement of existing techniques, e.g., speaker-dependent and independent recognition, coding, synthesis, speaker identification, etc.; (3) development and integration of knowledge-based techniques; and (4) development of speech assessment technology. These projects put emphasis on the multilinguism of recognition and synthesis. Multilinguism is considered to be prerequisite for the success of products on the international market. Some projects have just started; others are almost completed. For example, the project "Linguistic Analysis of European Languages" will reach completion at the end of 1988 and has built a knowledge base containing the necessary rules to solve ambiguities in phoneme-to-grapheme and grapheme-to-phoneme conversion for eight European languages. In addition, and by way of example, synthesizers and recognizers have been built for French and Italian, and a speaker verifier. This work will continue to be strongly encouraged in the future.

9:30

GGG6. Speech recognition in the French national project on man–machine interface. Jean-Paul Haton (CRIN/INRIA-Nancy, B.P. 239-54506 Vandoeuvre, France)

Research and development activities in the field of automatic speech processing have been relatively important in France for the past 20 years. In 1981, the Ministry of Research created the GRECO-PRC on Man–Machine Communication with the role of promoting and coordinating research activities in the fields of vision, natural language, and speech. This paper gives an overview of this national project which presently groups more than 40 French laboratories. First, its role, objectives, and means of action will be briefly introduced and then a description of the ongoing projects in the different fields that have been selected will be presented: (1) design, recording, and exploitation of a database of spoken French, BDSON, to be used for fundamental studies in acoustic phonetics as well as for assessment of speech recognition and synthesis systems; (2) morphological database of speech including knowledge-based techniques; (3) man–machine dialogue using speech; and (4) use of prosody in automatic speech recognition. The results already obtained will be presented and the future of the project will be discussed.

9:50–10:00
Break

10:00

GGG7. A joint Dutch research program for developing a high-quality text-to-speech synthesis system. Louis C. W. Pols (Institute of Phonetic Sciences, University of Amsterdam, Herengracht 338, 1016 CG Amsterdam, The Netherlands)

Since December 1985, a Dutch national research program has been in operation aimed at the realization of a laboratory prototype system for high-quality, unlimited-text-to-speech synthesis-by-rule for the Dutch language. Four phonetic laboratories (Amsterdam, Leiden, Nijmegen, and Utrecht), plus the Institute for Perception Research IPO in Eindhoven, and the PTT Research Laboratory in Leidschendam work together in this SPIN program. About 12 subtasks are defined ranging from defining the optimal structure of spoken text, grapheme-to-phoneme conversion, and morphological parsing, to studying speaker characteristics, intonation contours, and spectro-temporal speech parameters as a function of stress, speaking rate, and local context. Both allophones and diphones are used as basic units. Formal methods to evaluate the segmental and suprasegmental speech quality are developed and applied during the various phases of realization. In the final 2 years of the project, an optimal system architecture will be developed in order to integrate the single components and in order to achieve an operational implementation. [Work supported by SPIN, the Dutch National Program for the Advancement of Information Technology.]

10:20

GGG8. Cooperative speech technology projects—The Swedish perspective. Björn Granström (Department of Speech Communication and Music Acoustics, Royal Institute of Technology, S-100 44 Stockholm, Sweden)

Speech research has historically held a very strong place in Sweden, especially considering its small population of less than 9 million inhabitants. However, the industrial base for speech technology development is rather small. This is also true for many other countries in Europe. The multitude of small national languages asks for a wide spread of research activities on language specific topics. Cooperation both in research and speech technology product development has recently been started to overcome some of these problems. Sweden, which is still not part of the European Community, has until now been to some extent restricted in that work. The advent of ESCA, the European Speech Communication Association, and the new possibility for noncommunity countries to take part in, e.g., ESPRIT projects mean that Sweden now is taking full part in the growing European cooperation in this area. Some of that work will be described and some cooperative projects between the Nordic countries will be presented. A national Swedish program related to information technology development will also be discussed. This program is financed jointly by The Swedish Board for Technical Development and The Swedish Council for Research in the Humanities and Social Sciences. This acknowledgment of the nature of the subject area will stimulate new formations of cross-disciplinary cooperation.

10:35

GGG9. Voice input output for the K.B.C.S. project of India. P. V. S. Rao (Tata Institute of Fundamental Research, Bombay 400005, India)

Financed by the Government of India in cooperation with the United Nations Development Program (UNDP), the Nodal Centre for Speech Input Output is concerned with the realization of a voice-oriented interactive computing environment (VOICE) under a knowledge-based computer system project. The VOICE system aims at speech input backed by provisions for (a) feedback by (i) oral repetition by the machine, (ii) visual display; and (b) correction by (i) repetition of the word, (ii) spelling by voice, (iii) typing in. Speech output of the machine, based on an earlier work [P. V. S. Rao and R. B. Thosar, IEEE Trans. ASSP **ASSP-22**, 217–225 (1974)] will be either (a) a well-formed sentence based on a restricted syntax, or (b) the speaker's utterance, recognized, resynthesized, and repeated for confirmation. For (a), predetermined intonational patterns are used. For (b), the speaker's intonational pattern is superimposed on the synthetic utterance. For recognition, conventional subword unit approaches, hidden Markov models, and rule-based direct phonetic feature analysis techniques are all being evaluated. It is hoped that the final system will in some sense integrate the best features of these.

10:55

GGG10. Overview of speech research in Italy: National and European projects. Pietro Laface (Universita' di Salerno, 84081 Baronissi (SA), Italy and Politecnico di Torino, Corso Duca degli Abruzzi 24, 10129 Torino, Italy)

A survey of speech research that is being conducted in Italy, both in academic and in industrial institutions, is presented. Several groups are currently working on speech technology and research in Italy, but no efforts to integrate their activities toward common goals have been so far pursued mostly because no nationwide project has been organized and financed to promote their cooperation. Few commendable exceptions can be listed for which the collaboration between national groups has produced satisfactory results. The support for large-scale man–machine vocal interfaces programs has been granted instead by a number of European projects. Italian groups are participating in these projects in close cooperation with industrial and academic partners of different countries. Among these programs, probably the best known is the ESPRIT project, which is now at the beginning of its second phase. All the aspects of speech and natural language processing have been addressed, from speech analysis and synthesis to speech recognition system assessment; some of the most relevant results will be presented along with some proposals for perspective research. Another European EUREKA project in which Italian groups are involved is PROMETHEUS, an acronym for PROgraM for a European Traffic with Highest Efficiency and Unprecedented Safety, where processing of noisy speech, speaker independence, and dialogue systems are the main topics in the man–machine communication area.

11:10

GGG11. Speech recognition research at AT&T Bell Laboratories. Lawrence R. Rabiner (Speech Research Department, AT&T Bell Laboratories, Murray Hill, NJ 07974)

At AT&T Bell Laboratories, a broad range of systems for speech recognition, depending on the intended application area, has been studied. These systems have been extensively studied for isolated word (and phrase) recognition for command and control applications where a single word (or phrase) suffices to effect some type of control over a system or service. Typical applications include call type recognition for automated dialing and call routing within an organization. Several systems have also been implemented for connected word recognition with vocabularies ranging from the digits up to several hundred words. Typical applications for this technology include order entry, credit card entry, and digit dialing. Most recently, several studies have been begun of large vocabulary (1000–50 000 words), continuous speech recognition. The intended application for this type of system is database management and access. Our primary focus has been in studying alternative representations of subword speech units. In particular, recognition systems based on phones (acoustic–phonetic units), diphones, and acoustically defined units have been implemented and studied. Several methods of representing and accessing words from subword units including lexical access from a stored dictionary, explicit network representations of words, and derived acoustic lexicons have been studied. An investigation using a parser and a covering grammar to represent the constraints of the language and tack on the spoken input has also been shown. In this talk, the progress of each of these areas of speech recognition research is illustrated. Also discussed are the long-term research goals and the steps intended to take to reach these goals.

11:25

GGG12. Overview of telephone interpretation research at ATR. Akira Kurematsu and Kiyohiro Shikano (ATR Interpreting Telephony Research Laboratories, Twin 21 MID Tower, 2-1-61 Shiromi, Higashi-ku, Osaka, 540 Japan)

The Automatic Telephone Interpretation system is a facility that enables a person speaking in one language to communicate readily by telephone with someone speaking another. Basic research on the component technologies, speech recognition, machine translation, and speech synthesis is being carried out at ATR. In speech recognition research, the goal is to recognize continuous speech containing a large vocabulary of approximately 3000 words. Based on phonemes as the subunit of speech, words and phrases are recognized. The main approach is a hybrid combination of feature-based segmentation and phoneme and word recognition by hidden Markov and neural network models. A large-scale speech database with phonetic transcriptions has been developed for use in speech research. The incorporation of speaker adaptation has been chosen as our approach to speaker independence. Recognition of conversational speech on a specific domain will be explored. Integrating speech and language processing is an important area to be tackled. A technology that uses information about languages and background knowledge to understand the speech content is being studied. As for speech synthesis, rule-based speech synthesis with high quality and high naturalness is under development. In the area of machine translation, several approaches to the translation of spoken dialogue between Japanese and English are under study. Intention extraction for dialogue interpretation, a plan recognition model for intersentential analysis, and knowledge processing in a specific domain are intended to be investigated.

11:40

Panel Discussion: "Future Directions and Cooperation"

PANELISTS: Hiroya Fujisaki, *Moderator*
V. W. Zue
J. J. Mariani
P. V. S. Rao
L. R. Rabiner

FRIDAY MORNING, 18 NOVEMBER 1988 — WAIANAE ROOM, 8:00 TO 10:24 A.M.

Session HHH. Structural Acoustics and Vibration VII: Acoustic Absorption by Panels

Wayne T. Reader, Cochairman
David Taylor Research Center
Code 1905.2
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Jun'ichi Kanazawa, Cochairman
Kobayasi Institute of Physical Research
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Contributed Papers

8:00

HHH1. A method for measuring the acoustical reflectivity of coated panels at oblique angles of incidence. J. J. Dlubac and W. T. Reader (Code 1905.2, Ship Acoustics Department, David Taylor Research Center, Bethesda, MD 20084-5000)

Conventional short-pulse techniques for measuring the reflectivity of acoustic panels require that the reflected and incident waves be separated in time so that their ratio can be established. This technique is limited to moderate angles of incidence, since at high angles the direct and reflected waves overlap. An interference method is investigated as a way of making reflectivity measurements at high oblique angles. The total field (incident and reflected) in the vicinity of an oblique panel is measured relative to the incident field without the panel. The total field, which depends on frequency and angle of incidence as well as the panel material, is then examined and manipulated to extract information on the reflected wave.

8:12

HHH2. Complex impedance measurements in guided wave tubes. R. Harrison (Code R31, Naval Surface Warfare Center, Silver Spring, MD 20903-5000)

A pulsed guided wave tube operating over 2 decades of frequency was constructed for the measurement of the acoustic properties of materials. The impedance (Z) is calculated from the coefficient of reflection and the difference in phase shift between waves reflected from a steel reference plate and waves reflected from the material. The phase shift is obtained from the angle formed by the real and imaginary components of the Fourier transform of the reflected waveforms. Additional points for plotting in the Z plane are found by interpolation between the data points defined by the Fourier transform zeros in the reflection coefficient-phase angle space. Well-formed trajectories in the Z plane can now be plotted. Using the FFT has the advantage that (1) only a few accurately known data waveform points are needed as compared to traditional techniques that require a more complete waveform description of the waveforms to be compared and (2) the storage of data for calibration is facilitated. The entire system has been completely automated. Data will be presented showing the change of impedance with frequency for silicone rubber.

8:24

HHH3. Design of a three-layer acoustic window made from plastic. Toshiyuki Nakanishi (Japan Marine Science and Technology Center, Yokosuka, 237 Japan), Toshiaki Kikuchi, Akio Hasegawa, and Sumio Takahashi (National Defense Academy, Yokosuka, 239 Japan)

Theoretical and experimental studies are conducted on the design of an acoustic window for the acoustic imaging sonar to be equipped on "SHINKAI 6500." Frequencies used for this sonar are 100 kHz for PPI and 300 kHz for acoustic imaging. This acoustic window made from plastic material of good physical strength and low transmission loss at these frequencies is being developed. Generally, a single layer plastic plate has difficulty since it has both the longitudinal and the transverse waves in itself in the case of oblique incident acoustic waves in liquid, and, consequently, the transmission loss increases remarkably under the influence of these waves. The control for reducing the losses for the required frequency bands is accomplished by fitting matching layers on both sides of the plastic plate that constitutes the main layer. From calculations based on the theory, adequate acoustic velocity and thickness for the matching layers can be successfully obtained.

8:36

HHH4. Low-frequency reflectivity and transmissivity of gradual transition tank linings. John P. Tanzosh and Wayne T. Reader (David Taylor Research Center, Bethesda, MD 20084)

Anechoic chambers and tanks that are intended to perform over a decade or greater frequency range typically employ a gradual transition lining consisting of pyramidal sections, wedgelike sections, or triangular sheets of porous or voided material. The purpose of these tapered pieces of absorptive material, of course, is to present a gradually changing impedance to the sound waves incident upon the wall, while simultaneously conserving absorptive material from that required for continuous layers. The reflectivity and transmissivity of a lining consisting of a periodic array of arbitrarily shaped geometric sections are computed assuming that the acoustic wavelength in the fluid surrounding the absorptive sections is much greater than the spacing of the sections. Then, the sections and fluid between them are characterized by an arbitrary number of layers with an effective density and compressibility. Examples will illustrate use of the analysis to determine the shape and height of pyramidal and triangular sections of a voided viscoelastic polymer required to achieve specified performance goals.

8:48

HHH5. Dynamic characterization of materials: Error analysis of instrumentation and sample mounting configurations versus sample dynamic measurements as a function of material stiffness and form factor. Robert D. Collier and Jean Philippe Laures (Department of Mechanical Engineering, Tufts University, Medford, MA 02155)

The purpose of this investigation is to determine the relative effects of various sample mounting configurations on material measurements for characterization of dynamic modulus and loss factor. Measurements were performed on the Metravib viscoanalyzer over a frequency range up to 1 kHz. The results are compared for samples in both tension-compression and three-point bending configurations. For very soft rubbers, the relative stiffness characteristics of structural mounting and samples limit the range of measurements to below 500 Hz. Loss factors are consistent for both test configurations while dynamic modulus errors for three-point bending are dependent on mounting method and sample form. Optimum test configurations are defined. For various metals and composites, the limits of the two test configurations and optimum sample types and test configurations are determined.

9:00

HHH6. Measurement of Young's modulus of rubber in the high-frequency range. Toshiaki Kikuchi, Taiji Mori, Sumio Takahashi, and Akio Hasegawa (The National Defense Academy, Yokosuka, 239 Japan)

There is a close relation between Young's modulus of a medium and the oscillation behavior of a cavity in the medium. Young's modulus of polyurethane rubber was measured in the high-frequency range by a new technique. A measurement of the transmission loss of a rectangular rubber plate ($1\times20\times20$ cm) with spherical cavities in a water tank was made. It was shown that when the center-to-center spacing between the cavities was much larger than the diameter of the cavity, there exists a frequency at which the plot of the transmission loss versus frequency attains a maximum. This maximum loss frequency occurs when the circumference of the spherical cavity is equal to two shear wavelengths in rubber. Thus Young's modulus for rubber from the maximum loss frequency was obtained. It was shown that Young's modulus for rubber in the high-frequency range increased with frequency in the same manner as in the low-frequency range.

9:12

HHH7. Chlorosulfinated polyethylene: A versatile polymer for acoustical applications. Wayne T. Reader, Richard J. Deigan (Code 1905.2, David Taylor Research Center, Bethesda, MD 20084), and Robert W. Megill (E. I. Dupont De Nemours and Co., Inc., Wilmington, DE 19898)

Viscoelastic polymers are used extensively in the various fields of acoustics to reduce structureborne, airborne, and waterborne sound fields. Applications include their use as free or constrained layers adhered to vibrating plates to dissipate vibrational energy before it can generate noise in the surrounding field or can cause fatigue failure; as open cell foams applied to the walls of air chambers to absorb airborne sound before it can be reflected from the walls; and as closed cell foams that can be used to line the walls of water-filled tanks to render them nonreverberant and quiet. Each application requires that the elastic modulus and loss factor assume unique values within the operating frequency and temperature ranges. Chlorosulfinated polyethylenes, such as the HYPALON® family developed by E. I. Du Pont De Nemours and Co., possess the versatility needed to provide these unique properties and to be fabricated in the forms required for these acoustical applications. Discussed will be the range of elastic moduli, loss factors, and glass transition temperatures achievable by varying the HYPALON® polymer formulation, the cross-link density, and the amount and type of filler such as carbon black, oil, and type of blowing agent.

9:24

HHH8. Self-calibrating apparatus for the measurement of dynamic compressibility. David E. Edmonds (Applied Research Laboratories, University of Texas at Austin, Austin, TX 78713-8029) and A. W. Nolle (Department of Physics, University of Texas at Austin, Austin, TX 78712)

An apparatus has been developed to measure the complex dynamic compressibility for a sample without depressurizing the test chamber. Traditional measuring devices [McKinney *et al.*, J. Appl. Phys. **27**, 425–430 (1956)] require three measurement conditions per data point: chamber empty, containing an unknown sample, and containing a calibration sample. These chambers must be reassembled and repressurized for each condition, thus introducing two sources of error: the reassembly process, and the effect of pressure history on the internal piezoelectric transducers. The new apparatus employs a partition that allows the sample to be acoustically "added to" and "taken out of" the chamber. The chamber can be used to test organic solids in a range of 0–70 MPa (10 000 psi), below 0–60 °C, and up to 1200 Hz. Measurements at higher frequencies are feasible when corrections are made for acoustical inertances and resistances. Results for poly(vinyl acetate) have the same pressure dependence, and are the same in magnitude (within 7%) as those for a sample (though of different molecular weight distribution) measured by McKinney and Belcher [J. E. McKinney and H. V. Belcher, J. Res. Natl. Bur. Stand. Sect. A **67**, 43 (1962)].

9:36

HHH9. Measurement of loss factor by observation of phase shift and noise reduction of panels by damping. Jun'ichi Kanazawa and Jun'ichi Yoshimura (Kobayasi Institute of Physical Research, 3-20-41 Higashimotomachi, Kokubunji, 185 Japan)

The measurement of a decay curve or a frequency response is usually used to measure the loss factor of panels. Using an FFT analyzer, the phase shift was observed near the resonant frequency in this study. The loss factors measured from the phase shift and other methods were in good agreement except for the cases of higher loss factors at frequencies of higher modes of the panel. Theoretically, the loss factor is inversely proportional to the radiated sound power if the signal is random. In this study, the sound-pressure levels radiated from various panels of the same size and different loss factors were measured. They agreed fairly well with the predicted values, but the results for the panel with very low damping exhibited discrepancies from the theoretical values. This was due to the influence of the internal impedance of the panel as a vibrational source.

9:48

HHH10. Sensitivity analysis of Polymer Laboratories' dynamic mechanical thermal analyzer. R. J. Deigan and J. J. Dlubac (Code 1905.2, Ship Acoustics Department, David Taylor Research Center, Bethesda, MD 20084-5000)

Polymer Laboratories' dynamic mechanical thermal analyzer (DTMA) measures the dynamic Young's or shear modulus and loss factor of viscoelastic materials. The DMTA infers the modulus and loss factor by measuring the response of a small sample to forced vibrations in a small material sample. An analysis of the parameters affecting the DMTA dynamic modulus and loss factor data is presented. Test results are affected by errors in the input parameters or by violating the assumptions leading to the solution. The extent to which either error affects the test results is quantified. As a result of this analysis, recommended DMTA test procedures are presented.

10:00

HHH11. Acoustic coatings for a large water-filled tank. Robert D. Corsaro, Joel Covey, Brian Houston (Naval Research Laboratory, Code 5135, Washington, DC 20375-5000), Gregory Spryn, Paul Bednarchik, and Duane Weaver (Sachs/Freedman Associates, 1401 McCormick Drive, Landover, MD 20785-5396)

This paper describes four sound absorbing coatings designed to reduce extraneous wall echoes in water-filled pools in the frequency range 7.5–25 kHz. These coatings were specifically developed for use in the NRL Acoustic Pool Facility, where acoustic studies have occasionally been hampered by reflections from the concrete pool walls. At frequencies above 25 kHz, the pool is sufficiently large (11.5×8.0 m and 6.1 m deep) that wall echoes can generally be excluded from the data record by using familiar time-domain filtering; however, this filtering is not helpful for many interesting test geometries at frequencies below 25 kHz. Hence, to extend the usable frequency range of this facility, echo reducing wall coatings were considered. The most economical treatment meeting these requirements was found to require the use of four anechoic (antireflective) coatings, each with different characteristics and each covering selected regions of the pool. These four coatings were subsequently developed from classical design principles, using either layers or wedges of absorptive rubber. The acoustic characteristics of the materials used are also discussed.

10:12

HHH12. Hilbert transform method for determining the reflecting surface of acoustic materials. Yasushi Miki (Faculty of Engineering, Takushoku University, 815-1 Tatemachi, Hachioji, 193 Japan)

A new method for measuring delay time is proposed. If an observed signal is regarded as an output signal of a minimum phase system with delay, its delay time can be estimated by use of the Hilbert transform relationship between the log magnitude and the phase of the transfer function of the minimum phase system. Discussions are mainly focused on practical cases were the observed signal is bandlimited. A procedure for extrapolating the frequency characteristics of the signal is also presented to minimize the estimation error due to the bandwidth limitation. The method is applied to measure the boundary location of acoustic materials whose reflection characteristics are minimum phase. An estimation error less than ± 0.8 mm in distance is achieved when a test signal of 5-kHz bandwidth is used. As an example, the location of the reflecting surface of gravel is estimated. Using this result, the acoustic impedance of gravel is computed.

FRIDAY MORNING, 18 NOVEMBER 1988 — MAUI ROOM, 8:00 TO 10:37 A.M.

Session III. Underwater Acoustics VIII: Seismo-Acoustics of the Pacific Basin

N. Ross Chapman, Cochairman
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Victoria, British Columbia V0S 1B0
Canada

Tomoyoshi Takeuchi, Cochairman
University of Electro-communications
1-5-1 Chofugaoka
Chofu, 182 Japan

Chairman's Introduction—8:00

Invited Papers

8:05

III1. Ultralow- and very-low-frequency seismic and acoustic noise in the Pacific. John A. Orcutt (Institute of Geophysics and Planetary Physics (A-025), Scripps Institution of Oceanography, La Jolla, CA 92093)

A discussion of the sources of seafloor noise is most conveniently broken into four frequency bands since the noise is dominated by different physics in each of these bands. The first, and most familiar, band is from 3–50 Hz. This band is known as the *very-low-frequency* (*VLF*) or *infrasonic* band but is termed *high-frequency noise* for these puposes. The best documented mechanisms for the generation of ambient noise in this band is shipping. The next lower band, from 80 mHz to 3 Hz, is commonly called the *microseism band* after the high-level microseismic noise that is clearly recorded at all sites on the Earth's surface and results from nonlinear wave–wave interactions. The third band, the *noise notch* (20–80 mHz), has a variable bandwidth and is observed on both the continents and in the ocean. Noise levels within this notch appear to be controlled largely by currents and turbulence in the seafloor boundary layer. The final *ultralow-frequency* (*ULF*) band extends from dc to 20 mHz and the levels can be attributed to surface gravity waves. Very limited pressure and inertial displacement measurements have been made in the Pacific within these bands.

8:25

III2. Estimation of sediment shear Q using horizontal component OBS refraction data. Peter D. Bromirski, Frederick K. Duennebier, and L. Neil Frazer (Hawaii Institute of Geophysics, University of Hawaii, Honolulu, HI 96822)

Shear Q was estimated by amplitude comparison of horizontal component reflectivity synthetic seismograms with exceptionally high-quality horizontal component ocean bottom seismometer (OBS) refraction data collected at DSDP site 581C in the Northwest Pacific using a Soviet air gun source. Computations were performed on a CRAY X-MP/48 using an algorithm developed by Mallick and Frazer (1987). Sediment modeling was constrained by well log data and two-way V_P travel time from a normal incidence reflection profile. The velocity structure of the sediments and the upper 2 km of the basement was refined using ray tracing and reflectivity modeling. A strong shear conversion at the top of layer 2 constrains the shear-wave travel time through the sediments to about 1.8 s, giving an average V_S of about 0.2 km/s. Additional constraints on the modeling were imposed by matching the amplitudes of the primary as well as later arrivals. Preliminary results of a sensitivity analysis of the modeling indicate a minimum sediment shear Q of about 200 ($\alpha = 0.0136$ N/wavelength) at 9 Hz is necessary to match amplitudes of later arrivals. A stability analysis to determine the sensitivity of the modeling to Q_P, Q_S, density, and velocity will be presented. [Work supported by ONR.]

8:45

III3. Geoacoustic scattering from the Pacific seafloor. Ralph A. Stephen (Woods Hole Oceanographic Institution, Woods Hole, MA 02543)

It is amazing how much has been learned about the sediments and igneous crust beneath the sea by invoking the assumption of lateral homogeneity because almost every record obtained of sound propagating through the seafloor contains evidence of lateral heterogeneity and scattering. Three scales of scattering are evident: (1) There is the focusing and scattering of energy due to lateral heterogeneity within the seafloor. This causes anomalously high- or low-amplitude arrivals but no new arrivals; (2) there are diffractions from isolated topographic irregularities such as hills and valleys. In some cases, the hyperbolic appearance of these arrivals is evident on refraction sections; and (3) fine scale heterogeneity in the seafloor causes incoherency and time spread of arrivals. These are ubiquitous effects in seafloor reflections and refractions. These three phenomena will be demonstrated in observed data from the Pacific and in synthetic seismograms.

9:05

III4. Development of off-line, deep-towed seismic profiler. Kiyokazu Nishimura, Kiyoyuki Kishimoto, Teruki Miyazaki, Masato Joshima, and Fumitoshi Murakami (Marine Geology Department, Geological Survey of Japan, 1-1-3 Higashi, Tsukuba, 305 Japan)

A deep-towed seismic reflection profiler has been developed by the Geological Survey of Japan. The aim of the system is to obtain detailed profiles of subbottom structures in the deep seas. The system consists of deep-towed seismic recording packages with a deep-sea hydrophone streamer, a shipboard data logging and processing system, and an acoustic navigation system for the deep-towed packages. The packages are towed near the seafloor, and a sound source (an air gun) is towed near the sea surface. After the recovery of the packages on board, the seismic data in the package are transferred to the data logging and processing system through a high-speed serial link. The system has the following features. (1) Off-line towing: the hydrophone streamer and the recording packages are towed by an ordinary wire without a conducting cable. (2) No depth limitation: all electrical packages are kept in pressure-proof housings (depth limit 10 000 m), and unlimited-depth hydrophone elements are used for the streamer. (3) Digital recording and nonvolatile, solid-state memories: an A/D converter with a floating-point amplifier is used. Digitized seismic data are recorded in 16-megabyte magnetic bubble memories. Field tests of the system were successfully carried out. Some preliminary profiles of deep-towed digital seismic data were obtained.

Contributed Papers

9:25

III5. Seismic profiling at the West Pacific Ocean by DDC1-1 geosonar. Zhang Shuying and Ren Leifa (Shanghai Acoustics Laboratory, Academia Sinica, 456 Xiao-Mu-Qiao Road, Shanghai, People's Republic of China)

Some valuable records of seismic profiling at the West Pacific Ocean have been obtained by using DDC1-1 geosonar. A significant discovery is that a layer of sediment, the thickness of which is more than 500 m, exists at the area of Ryn-Kyn Ocean Trench, where the water depth is over 6700 m. The DDC1-1 geosonar consists mainly of a 30 000-J sparker as a strong wideband sound source and a 50-m-long streamer, including 20 equally separated hydrophones and an echo-processing system. Several new techniques have been developed for the improvement of both penetration and resolution of seismic profiling, such as: (1) time-varying filtering—the frequency band of the sonar receiver can be varied in real time with the variation of the echo travel time according to an adjustable pattern depending on the acoustic attenuation of marine sediments; and (2) synthetic-aperture processing—an exponential accumulator is used for stacking a series of echoes from some certain subsurfaces of sediments in real time corresponding to successive shots of the sound source. The gain of echo to interference ratio by means of this technique is determined by

$$G = 10 \log \left[(2N-1) \frac{1-(1-1/N)^p}{1+(1-1/N)^p} \right] \text{dB},$$

where p is the stacking time and N is an adjustable parameter that determines the maximal gain of the accumulation processing.

9:37

III6. Low-frequency seismoacoustic ambient noise levels. A. C. Kibblewhite and C. Y. Wu (Department of Physics, University of Auckland, Auckland, New Zealand)

Studies of the infrasonic ocean noise induced by the interaction of surface waves have shown that the noise levels are dependent not only on the ocean-wave field but also on the geoacoustic properties of the environment. A full understanding of the processes involved requires an examination of the role of the inhomogeneous components of the induced acoustic field and the reflection coefficient of a multilayered seabed. An analysis has shown that, at infrasonic frequencies, the inhomogeneous component can be important in waters with depths less than a few hundered meters. It has also been established that the reflection loss for particular models shows energy conversion from interface waves to trapped normal modes, when a low-speed layer is present within the geoacoustic sequence. This conversion leads to an apparent periodicity in the frequency dependence of the reflection loss.

9:49

III7. Estimation of geoacoustic models of the seabed from bottom loss measurements. N. R. Chapman,[a] R. W. Bannister (Defence Scientific Establishment, Auckland, New Zealand), and R. Falconer (GeoResearch Associates, Waikanae, New Zealand)

Measurements of bottom loss obtained in deep water sites in the S.W. Pacific have been used to infer geoacoustic models of the ocean bottom. The measurements were grouped into two types according to the acoustic reflectivity at frequencies below 100 Hz. The first type was characteristic of deep ocean basins and the second was typical of shallower regions of the continental margin. This classification was supported by archival seismic and nonacoustic information that indicated that the geology and the geophysical properties of the sediment were significantly different for each type. The former was composed of a layer of silty-clay material overlying the ocean crust, whereas the latter consisted mainly of sandy turbidite layers. A ray model, which was developed to simulate bottom loss in thin sediment layers, was used to estimate values of the geoacoustic properties for each bottom type by comparing model results with the measurements. The data at low frequencies were adequately described by simple geoacoustic models consisting of a layer of sediment with a constant sound-speed gradient overlying a solid substrate. For the first bottom type the substrate material was basalt, whereas for the second type the substrate was consolidated sediment. Results are presented for an example of each bottom type. [a] On exchange from Defence Research Establishment Pacific, FMO, Victoria, B.C. V0S 1B0, Canada.

10:01

III8. Towed array measurements of bottom time/angle spreads during Pacific Echo. Ronald L. Dicus (Naval Research Laboratory, Code 5120, Washington, DC 20375)

Pacific Echo was a joint United States/Canadian experiment conducted in the Pacific in June 1986. As part of the exercise, explosive charges (MK-61 SUS) were detonated along a closing-range track near broadside to a towed array. A high-level source (MK-VI, 193 dB at 15 Hz) transmitted linear frequency-modulated signals, sweeping from 5–15 Hz and back in 50 s. Measurements reported here are for a thin-sediment site just south of the Tufts Abyssal Plain. Previous omnidirectional measurements in thin-sediment areas are characterized by bottom time extensions on the order of a second. Time (angle) spread is defined here by identifying a two-dimensional area in time/angle space containing an arrival, integrating over angle (time), convolving the resulting one-dimensional sequence with a rectangular window, and determining the length of the window giving a peak -3 dB down from the total energy of the particular arrival. Observed time and angle spreads increased with grazing angle and frequency to values on the order of 0.5 s and 15° over a frequency and grazing angle range of 10–300 Hz and 7°–70°, respectively. The dependence of reflectivity, time/angle spreads, and signal gain on frequency and bottom grazing angle will be discussed.

10:13

III9. Very-low-frequency scattering by abyssal hills of the Pacific Ocean. O. Diachok and E. Livingston (Naval Research Laboratory, Code 5120, Washington, DC 20375-5000)

Tectonic forces operating on the upper crust over its first million years (within about 20 km of the crest of the East Pacific Rise, EPR) give rise to a "horst and grabben" structure that hypothetically remains frozen as the plates advance across the Pacific Ocean. The horsts (uplifted plateaus) and grabben (valleys) striate the Pacific bottom in a nearly north–south direction. Scripps deep tow measurements at a nearly sediment-free site about 30 km from the EPR reveal that the number, height, and width of the horsts are nominally 2/km, 60 m, and 600 m, respectively. Twersky's theory of scattering from elliptically striated surfaces appears to be appropriate for estimating/investigating scattering loss and modal stripping caused by this type of bottom. Examples of calculations will be presented at infrasonic frequencies. The resultant computations constitute real world predictions provided that the acoustic wavelength exceeds the thickness of the sediments overlaying the basalt. In the majority of the Pacific, the nominal sediment thickness is less than 100 m and the corresponding frequency range is below 15 Hz.

10:25

III10. Project SWATHMAP: Military sonars in service to science. Peter B. Humphrey (Department of Geology and Geophysics, University of Hawaii Institute of Geophysics, Honolulu, HI 96822)

Project SWATHMAP (one of four deep-water, long-range sidescan sonars in operation today) is the low-cost geologic application of a U. S. Navy antisubmarine warfare system utilized on routine ocean-wide deployments. While resolution is not sufficient to observe bathymetric structures in detail, the system is particularly adept at locating them, discerning scale and shape, and determining continuity. Routine observations include terraces, trench crossings, seamounts (many of them newly discovered) often topped by craters, fracture zones, and abyssal hills (both superb clues to plate tectonic motion). Discoveries on this scale are essential for ground truthing the next decade's satellite altimeters. Certainly there are sidescan systems that do a better job of seafloor mapping than American combat vessels, but their cost is extraordinary by comparison and well beyond the reach of most nations. By contrast, even the humblest antisubmarine warfare-equipped navy can use such a system to locate most substantial features within its exclusive economic zone. Possible system upgrades include (1) two-sided scanning for maximum coverage, (2) stereo image production from parallel passes or sequential imaging, (3) digital tape recording for subsequent image manipulation and enhancement, and (4) decipherment of parallax across the array to determine the acoustic arrival angles one utilizes to estimate bathymetry.

Session JJJ. Physical Acoustics VIII: General (Poster Session)

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Contributed Papers

All posters will be displayed from 9:00 to 11:00 a.m. To allow all contributors the opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:00 to 10:00 a.m. and contributors of even-numbered papers will be at their posters from 10:00 to 11:00 a.m.

JJJ1. Simultaneous sensing of the acoustic wave and temperature using a polarization-maintaining fiber. Sumio Takahashi, Toshiaki Kikuchi, Ryohei Yagi, and Akio Hasegawa (Department of Applied Physics, The National Defense Academy, 1-10-20 Hashirimizu, Yokosuka, 239 Japan)

Many types of fiber-optic sensors have been proposed and developed. In general, efforts have been made to detect a specific parameter of interest separately from other parameters. In the homodyne interferometric type of sensors, the fading of output caused by temperature variation is a serious problem, and several types of compensation schemes have been developed. In contrast, this paper describes the simultaneous sensing of acoustic wave and temperature change using a polarization-maintaining fiber. The acoustic wave is detected by an optical heterodyne scheme and temperature change is detected by an optical homodyne scheme. The zeroth-order diffracted light from a Bragg cell is directed into both the X and Y polarization axes of a polarization-maintaining fiber. The homodyne interference of the X and Y components is utilized to detect the temperature variation. At the same time, the heterodyne interference of the X polarization component of the output light from the fiber and the first-order diffracted light propagated in air is used to detect the acoustic wave. Simultaneous sensing of the acoustic wave and temperature variation was successfully demonstrated by this method.

JJJ2. Elastic wave propagation in multilayered anisotropic media with applications to fibrous composites. Adnan H. Nayfeh (Department of Aerospace Engineering, M.L. 70, University of Cincinnati, Cincinnati, OH 45221-0070) and Dale E. Chimenti (Materials Laboratory, Wright–Patterson AFB, Dayton, OH 45433)

Theoretical analysis supported by extensive experimental investigations for the propagation of elastic waves in multilayered anisotropic media is presented. Each component of the solid is allowed to possess up to monoclinic symmetry properties. Furthermore, the wave is allowed to propagate along any azimuthal direction. Both free and forced waves are considered. Waves incident from within a supporting fluid on the fluid–multilayered medium interface are given special attention. Here, the reflection and transmission coefficients are derived from which all propagation characteristics are identified and are readily available. The analytical results are exact and hence cover all frequency ranges. Experiments are conducted on a multilayered fibrous composite plate of graphite epoxy lamina. Results are presented in the forms of comparisons of theory and experiments for dispersion characteristics. Also, comparisons of reflected energy distribution as a function of frequency times thickness are included.

JJJ3. Forward projection of a transient tomographically reconstructed pressure field. Michael J. Forbes,[a] Stephen V. Letcher (Department of Physics, University of Rhode Island, Kingston, RI 02881), and Peter R. Stepanishen (Department of Ocean Engineering, University of Rhode Island, Kingston, RI 02881)

The forward projection of transient pressure fields from ultrasonic transducers has been investigated both experimentally and theoretically. Previously developed impulse-response methods have been used to determine the space- and time-dependent pressure field of a particular plane in front of a typical ultrasonic transducer. The pressure field of this plane is then projected to another plane further from the transducer using wave-vector and time-domain methods that are implemented using FFTs. The numerical results illustrate the accuracy of the approach using the impulse-response method for purposes of verifying the method of projection. Experimentally determined transient pressure fields in a plane in front of typical medical or NDE transducers have been obtained using optical tomographic methods to measure a two-dimensional slice of the field as a function of time. The forward projection of the space- and time-dependent field via the projection algorithm yields results that are in good agreement with the optically measured field at the new plane. A comparison of analytical and experimental results will be discussed along with potential applications of the method for transducer calibration and characterization. [a] Present address: Naval Underwater Systems Center, Newport, RI.

JJJ4. Observation of impulse response for ultrasonic transducers by optical computed tomography. Hiroyuki Obara and Kimio Shibayama (Faculty of Engineering, Tamagawa University, 6-1-1 Tamagawa-gakuen, Machida, 194 Japan)

The present study is aimed at developing accurate and simple techniques to observe the behavior of ultrasonic transducers and their holders. To know the impulse response of wideband type transducers driven in water, it is important to investigate the ultrasonic field pattern under actual performance. One solution would be given by the combination of the optical CT and the impulse response analysis by the M-sequence correlation method. The experimental arrangement of the Raman–Nath diffraction was used. The signal was given by a PIN-photodiode, which included the responses of all the components of the Fresnel diffraction. The optical measurement system using the M-sequence correlation method gives high stability and sensitivity over a wide frequency range. The ultrasonic signal in the megahertz region was processed by a low-cost slow A/D converter with 12-bit resolution. The experiment was conducted with a 13-mm concave transducer, and the instantaneous wave front was reconstructed every 50 ns. The effect of the electrodes could be clearly observed.

JJJ5. Electrode shaping to achieve Gaussian ultrasonic field distribution. M. A. Breazeale, J. J. Wen, and J. Na (Department of Physics, The University of Tennessee, Knoxville, TN 37996)

The electrode shape required to produce a Gaussian electric field distribution at the surface of a transducer was calculated. Such electrodes were made and tried with both quartz and PZT transducers with promising results. These transducers have an advantage over those made by taking advantage of electrical fringing [Breazeale *et al.*, J. Acoust. Soc. Am. 70, 1791–1793 (1981)] in that they do not have the limitation $2 < W/T < 4$. Results of measurement of the ultrasonic field produced at 4 MHz by quartz and PZT are given, and application to NDE and acoustical microscopy is considered. [Research sponsored, in part, by UT-ORNL Science Alliance, A State of Tennessee Center of Excellence. PZT provided by S. Fujishima, Murata Corp.]

JJJ6. The collection efficiency of an electrostatic precipitator with ultrasonic application. Tomoo Nakane and Koichiro Seya (Department of Electrical Engineering, College of Industrial Technology, Nihon University, 1-2-1 Izumicho, Narashino, 275 Japan)

An electrostatic precipitator (ESP) has a lower collection efficiency for smaller particles. It is believed that increasing the submicron particles sufficiently for the operation of ESP would lead to a good collection efficiency. A high-intensity ultrasonic field was produced inside an ESP, and the collection efficiency was examined, and, as a result, a higher collection efficiency was obtained with the application of ultrasonic agglomeration. The collection efficiency obtained was higher than previous reports. The experimental ESP was of the pipe type and sample particles were made by diluted burning incense sticks in this experiment. Sound waves were produced by a stepped circular vibrating plate at a frequency of 20 kHz.

JJJ7. Visual observation of electrical discharges in a standing sound wave field. Takashi Hirata and Koichiro Seya (Department of Electrical Engineering, College of Industrial Technology, Nihon University, 1-2-1 Izumicho, Narashino, 275 Japan)

There are many papers reporting the effects of electrical discharge, such as in atmospheric pressure, gaseous mediums, and the shape and geometry of an electrode. However, in spite of the fact that an electrical discharge phenomenon accompanies sound waves, there are very few reports regarding the influence of sound waves on the discharge phenomenon. In this paper, it is visually confirmed that discharging phenomena are influenced by sound waves irradiating the electrical discharge. Discharging cells with both positive and negative needle electrodes were located at a sound-pressure node in a standing wave field. As a result, the luminous part of the electrical discharge changed to a leaflike shape, and the starting voltage of the discharge also changed.

JJJ8. Alleviation of frictional electrification of the DIP type IC with ultrasonic vibration. Koichiro Seya and Tomoo Nakane (Department of Electrical Engineering, College of Industrial Technology, Nihon University, 1-2-1 Izumicho, Narashino, 275 Japan)

There are some instances in which a DIP type IC is damaged by frictional electrification in the manufacturing process. To ground the IC is one of the means of protecting it from electrostatic damage, although it is not perfect. It is believed that decreasing the contact area between the DIP type IC and other objects is effective in reducing the charging of the IC. The effective area and the time of contact are expected to decrease if the contact point between the IC and the other object is vibrated. A DIP type IC was passed through a slideway vibrating at a frequency of 22 kHz. The charged level of the IC was found to decrease. The present method of applying ultrasonic vibration to reduce the electrification of the DIP type IC could be used in the manufacturing process.

JJJ9. Measurements of the acoustic characteristics of newly developed plastics. Masahide Gakumazawa, Seiji Kaneko (Department of Electrical Communication, Shibaura Institute of Technology, Minato-ku, Tokyo, 108 Japan), and Sadayuki Ueha (Tokyo Institute of Technology, Nagatsuta, Midori-ku, Yokohama, 227 Japan)

Newly developed plastics, including FRP, which are called engineering plastics, have been widely used in place of metals and are often welded together or machined by means of ultrasonic vibrations. However, their acoustic characteristics have been hardly reported, except for their physical properties. The attenuation constants and velocities of such plastics have been measured by using a simple pulse-echo method. The frequencies used were 0.5, 1.0, 2.0, and 5.0 MHz. The frequency dependence of the attenuation was measured for several kinds of plastics. The attenuation of engineering plastics was shown to be larger than that of conventional plastics.

JJJ10. Ultrasonic relaxation in GeO_2 glass. Keiji Sakai, Pak-Kon Choi, and Kenshiro Takagi (Institute of Industrial Science, University of Tokyo, Minato-ku, Tokyo, 106 Japan)

Ultrasonic velocity and absorption were observed in GeO_2 glass over the frequency range of 20 MHz–1.5 GHz. High-resolution Bragg reflection (HRB) technique was successfully utilized at the higher frequency range, where the conventional pulse method loses accuracy due to the very strong attenuation. A pulse apparatus equipped with a PVDF polymer transducer was useful over the 20- to 90-MHz range. It is known that GeO_2 has a large absorption maximum with respect to temperature if measurements are made in the MHz range. In the observed absorption spectrum, it was found that the corresponding maximum was centered at around 1 GHz at room temperature. The excess absorption was ascribed to a relaxation mechanism, in which the molecular structure was deformed in a double-well potential. The present results suggest that the height of the energy barrier between the two states has a distribution of the Gaussian type centered at 0.13 eV and with a width of 0.09 eV. Assuming this energy barrier, the ultrasonic absorption was predicted to be a function of temperature. The result of prediction was in good agreement with the present experiment made in the low-temperature region.

JJJ11. A method for ultrasonic manual testing and classification of test results for welds of ferritic steel with acoustic anisotropy. Hisao Yamaguchi (Sumitomo Metal Industries, Ltd., 1-3 Nishinagasu-Hondori, Amagasaki, 660 Japan)

Angle-beam ultrasonic testing has been applied to flaw detection in steel welds according to specifications based on the isotropy of the steel. However, certain types of rolled steel plates show an anisotropy that arises during the rolling processes necessary to satisfy the required mechanical qualities. In the angle-beam ultrasonic testing of steel with some degree of anisotropy, a change in the refraction angle and a decrease in the echo amplitude are observed. Considering this fact, the Japanese Society for Nondestructive Inspection has made a new standard for the angle-beam ultrasonic testing of welds of steel plates with anisotropy over 1.02 in shear wave velocity ratio. The following items are specified in the standard, which is mainly applied to welds of ocean structures and ships: (1) It is recommended that probes of a 60- or 65-deg nominal refraction angle be used. (2) The estimation of the flaw location should be made by using the refraction angle measured in the test steel. (3) The sensitivity should be adjusted using reference blocks with side drilled holes.

JJJ12. Dual-type helical acoustic antenna. Akio Hasegawa, Sumio Takahashi, Toshiaki Kikuchi, and Teruaki Fujita (Department of Applied Physics, The National Defense Academy, 1-10-20 Hashirimizu, Yokosuka, 239 Japan)

A dual-type helical acoustic antenna, which operates as an underwater endfire hydrophone, is described. It consists of two brass-wire helices wound in a concentric configuration and a PZT piezoelectric transducer of Langevin type. The structure is analogous to a helical antenna operating in the axial mode of radiation at microwave frequencies. The progressive wave along the helix yields the phase shift $k_b l$ due to wave propagation corresponding to the length of one turn l, while the phase shift due to the spacing between turns (the pitch) S is $k_w S$, where k_b and k_w are the wavenumbers in the brass wire and in water. For the radiation from the corresponding segments of all turns to add in phase in the axial direction, it is necessary for the helices to satisfy the conditions of

$k_h l_i = k_w S_i$ and $k_h l_o = k_w S_o$, where the subscripts of i and o represent the inner and outer helices. The relationship between directional patterns and the dual helix parameters is shown in detail.

JJJ13. Viscosity sensor using a surface acoustic wave delay line. Tooru Nomura, Syu Iino, and Tsutomu Yasuda (Department of Electrical Communication, Shibaura Institute of Technology, 3-9-14 Shibaura, Minato-ku, Tokyo, 108 Japan)

A surface acoustic wave (SAW) sensor for sensing viscosity is presented. SAW devices are commonly used in electronic signal processing. SAW delay lines offer many attractive features for applications as vapor and liquid sensors. In this paper, a SAW viscosity sensor utilizing the shear horizontal displacement of a leaky SAW mode is presented. The sensor is based on a dual SAW delay line oscillator. The leaky SAW with the shear horizontal displacement is excited on a 36°-rotated Y-cut X-propagation $LiTaO_3$ substrate using interdigital transducers. The sensitivity of the SAW oscillator frequency to the viscosity of liquids has been measured. It is shown that the viscosity sensor has a sensitivity of about 100 Hz/cP at an operating frequency of 35 MHz with a threshold sensitivity of about 2 cP. Test results to include response time, hysteresis, linearity, and temperature effects will also be presented.

JJJ14. Two-dimensional optical beam scanning using a leaky surface acoustic wave device. Heijiro Urabe and Kohji Toda (Department of Electrical Engineering, The National Defense Academy, Hashirimizu, Yokosuka, 239 Japan)

Two-dimensional (2-D) optical beam scanning is obtained by using a leaky surface acoustic wave device. The device consists of two interdigital transducers (IDTs) on one of the surfaces of a piezoelectric substrate, operating at a liquid–solid boundary. Ultrasound radiated into the liquid from each of two IDTs is valid for acousto-optic (AO) deflection gratings, one for horizontal and the other for vertical deflection. The fundamental performance of the AO deflector is characterized in the relation of the input electrical admittance of the leaky wave transducer with the specific acoustic impedance of the liquid by an equivalent circuit model analysis. The device has the linearity essential for a laser scanner in a wide frequency range. The 3-dB intensity area of the 2-D deflected spot is 0.8 deg for a frequency bandwidth of 30 MHz. The access time is 0.66 μs, and the reasonable spot numbers are 20×20 for an optical beam diameter of 1 mm.

JJJ15. Experimental investigation of the SAW harmonic reflection characteristics of a metal strip array. Toshihiro Kojima, Hiroyuki Obara, and Kimio Shibayama (Faculty of Engineering, Tamagawa University, Machida, 194 Japan)

It has been reported by R. C. M. Li and J. Melngailis [IEEE Trans. Sonics Ultrason. SU-22 (3), 189–198 (1975)] that a metal strip array on Y–Z $LiNbO_3$ shows a strong reflection of the surface acoustic wave (SAW) at the frequency of the second harmonic component due to the energy storage effect. To apply this properly to reflectors for a SAW resonator, the following characteristics for both shorted and open metal strip arrays on $LiNbO_3$ (128° rot. Y, X-prop.) have been experimentally investigated. The dependence of the second harmonic reflection coefficient on the metalization ratio and the comparison between the reflection coefficient for the second harmonic and the fundamental frequency regions have been examined in detail. The relations between the frequency at which the maximum reflection coefficient occurs and the center frequency of an interdigital transducer (IDT) have also been determined. The characteristics obtained here are useful for the optimum design of second harmonic, one- and two-port resonators.

FRIDAY MORNING, 18 NOVEMBER 1988 — WAIANAE ROOM, 10:30 A.M. TO 12:06 P.M.

Session KKK. Structural Acoustics and Vibration VIII: Structural Wave Propagation

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Contributed Papers

10:30

KKK1. Strict passbands of waves on flat, periodically supported, fluid-loaded, elastic layers. P. W. Smith, Jr. (BBN Laboratories Incorporated, Cambridge, MA 02238)

A "strict" passband is defined as a frequency interval in which the Floquet wave on a periodic surface propagates without any attenuation. Their existence is characteristic of conservative systems *in vacuo*. Examples of strict passbands on a system in contact with an infinite fluid have been published, but, in each case, the elastic layer is a classical thin plate. In such a band, energy flows along the wetted layer, but none is radiated as sound despite the supports. It is shown that strict passbands exist for any layer, however complex in the thickness direction, if the system satisfies certain broad conditions: (1) The system is conservative; (2) the supporting ribs do not cut the elastic layer; and (3) the layer is statically stiff at wavelengths in the neighborhood of twice the rib spacing.

10:42

KKK2. Evaluation of a procedure for the analysis of nonstationary random vibroacoustic data. Allan G. Piersol (Astron Research and Engineering, 3228 Nebraska Avenue, Santa Monica, CA 90404), Harry Himelblau (Jet Propulsion Laboratory, Pasadena, CA 91109), and Donald M. Wong (The Aerospace Corporation, Los Angeles, CA 90009)

Studies have been performed using space shuttle launch vibration measurements to evaluate a proposed procedure for the improved analysis of the nonstationary random vibroacoustic data acquired during space vehicle launches. The procedure involves decomposing the launch data into separate time- and frequency-dependent components by exploiting a parametric model (called the product model) that approximates the nonstationary character of the data. This decomposition permits the time dependence of the data to be analyzed over a wide frequency bandwidth, and the frequency dependence to be analzyed with a long averaging time, in both cases providing statistically stable results with a large bandwidth-averaging time (BT) product. Applications of the procedure to sample space shuttle launch vibration measurements are compared to the results of traditional analysis techniques based upon the computation of spectra over short time durations, where overall values are a maximum. The comparisons indicate that the proposed analysis procedure not only reduces the random errors in spectral estimates, but also suppresses the tendency to underestimate the maximum vibroacoustic loads during launch. [Work supported jointly by NASA and USAF.]

10:54

KKK3. Vibration damping in consideration of the directivity of vibration perception. Matunori Nara, Teruhiko Konishi, and Hiroyuki Okuno (Research and Development Center of DND Co. Ltd., 3-56-2, Nangai, Higashiyamato, 189 Japan)

The difference between vertical and horizontal vibration perception was found from consideration of the character of the receptor. The equivalent sense distribution of the vibration perception was measured at frequencies ranging from 0.5–20 Hz. The experiments were aimed at developing the optimal method for vibration damping. The relation between the frequency and the vibration level was found to be described exactly by the transfer function of a second-order integral system with delay with a pair of complex poles. Under constant frequency and the maximum vibration transmissibility, the ratio of the receptor's damping factor for the horizontal vibration to that for the vertical (at 0.0075) was 1.8. According to the relation between the impulse response and the damping factor, doubling the damping factor causes approximately 13% decrease in the impulse response. Therefore, it was found that the load of vibration on the human body could be decreased by converting a vertical vibration into a horizontal one.

11:06

KKK4. Resonant frequencies and mode shapes of open fluid-loaded prolate spheroidal shells. Jack Yahner (Aerospace Corporation, P. O. Box 92957, Los Angeles, CA 93339) and Courtney B. Burroughs (Applied Research Laboratory, The Pennsylvania State University, P. O. Box 30, State College, PA 16804)

Governing differential equations for the vibrations of open fluid-loaded prolate spheroidal shells are presented. Transverse shear and rotary inertia are included in the shell equations. The open end of the shell, defined by an axisymmetric coordinate line, is assumed to be clamped. Approximate solutions for the axisymmetric resonant frequencies and mode shapes of *in vacuo* and fluid-loaded shells are derived using Galerkin's vibrational method and are compared.

11:18

KKK5. Vibrations of damaged trapezoidal plates. David K. Holger (Department of Engineering Science and Mechanics, 2019 Black Engineering Building, Iowa State University, Ames, IA 50011) and Qunying Jiao (Department of Basic Science, Beijing Agricultural Engineering University, Beijing, People's Republic of China)

The modal frequencies and shapes of damaged edge-clamped trapezoidal plates have been investigated for the first five transverse modes. Damage considered consisted of narrow, rectangular, centered slots. An impulse hammer was used to excite transverse plate vibration, and plate response was measured with a nearfield microphone. The frequency response of the plate was investigated for all combinations of three slot lengths and three slot orientations. The first five modal frequencies and shapes were estimated from ensemble-averaged, frequency-response functions. Observed changes in modal frequencies for the damaged plates were in good agreement with predictions of a finite element model. The results suggest that slot presence can be detected from changes in plate modal frequencies, that slot length and orientation have significant effect on the modal frequencies, that a nearfield microphone is an adequate response transducer, and that an appropriate finite element model and the frequency response measurement are of comparable sensitivity to damage. It is suggested that combining experimental modal analysis and the finite element method has significant potential for macroscopic, nondestructive evaluation.

11:30

KKK6. In-plane and bending wave analysis of a step discontinuity in a thick plate structure. M. McCollum and J. M. Cuschieri (Center for Acoustics and Vibrations, Department of Ocean Engineering, Florida Atlantic University, Boca Raton, FL 33431)

Having developed the response equations for in-plane and bending waves in a thick L-shaped plate structure of infinite extent (Noise-Con 88), when both plates are of the same thickness and material, the analysis is now extended to plates of different thicknesses. In this study, instead of the L-plate configuration, a flat plate with a step discontinuity is investigated. The boundary between the two plates is excited by a Mindlin bending wave at some arbitrary angle of incidence. In this paper, the complete thick plate solution for the transmitted and reflected waves is obtained using Mindlin's equations to describe the plate flexure, and including the generation of in-plane shear and in-plane longitudinal waves at the junction. The solution is obtained for different angles of incidence and different step configurations. The results that will be presented are for the total transmitted power and the power transmitted by the individual wave components, averaged over all angles of incidence, as a function of the plate step discontinuity.

11:42

KKK7. Narrow-band random vibration of a van der Pol oscillator. Huw G. Davies, Qiang Liu, and D. Nandlall (Department of Mechanical Engineering, University of New Brunswick, Fredericton E3B 5A3, Canada)

The response of a van der Pol oscillator to random narrow-band excitation is considered when the excitation frequencies are either near to or away from the system natural frequency. The analysis is based on a multiple time scale perturbation series and stochastic averaging. Near the system frequency, the response is similar to the purely sinusoidal case. Away from resonance, the response can include an entrained response at the system frequency driven by a random parametric excitation. Time-dependent envelope equations agree well with numerical simulations. The envelope equations are used to find the response probability density functions for very small excitation bandwidths. [Work supported by NSERC Canada.]

11:54

KKK8. Acoustic guided waves in periodical axisymmetric structures. G. Ching and M. Lagier (DSA Thomson–Sintra ASM, 525, route des Dolines Parc de Sophia Antipolis, BP 38, 06561 Valbonne Cedex, France)

Acoustic wave propagation in axisymmetric periodic waveguides concerns many different application fields: acoustic waves in ducts, acousto-mechanic waves in bore hole, and noise propagation in towed arrays. A periodic guide with an intricate elementary cell and coupling with an external medium (possibility of leaky waves) makes the theoretical analysis of such waveguides more difficult. The present work is based on the combination of three analysis tools: finite element analysis for the elementary cell dynamics, Bloch–Floquet wave decomposition for the periodic structure processing, and the perturbation method applied to harmonic space in order to take into account the external medium interactions. Dynamic matrices partitioning and the introduction of input–output propagation conditions lead, in the case of zero coupling (with the external medium), to the typical secular equation: $[P+\Gamma Q+\Gamma^2 R]x=0$, where Γ is the unknown propagator and x is the associated eigenvector; P, Q, and R are matrices deduced from the initial dynamic matrices. The external medium boundary condition introduces an additional nonpolynomial term and leads to a new nonlinear secular equation that is difficult to solve. The perturbation method allows, on the one hand, this difficulty to be overcome and, on the other hand, leads to a more physical approach, particularly in the method state where coupling leads to a Bloch–Floquet harmonic wave degeneracy overtaking. Preliminary numerical tests on well-known waveguides are presented. The results agree very well with analytical solutions. [Work supported by GERDSM.]

FRIDAY MORNING, 18 NOVEMBER 1988 — MAUI ROOM, 10:40 A.M. TO 12:00 NOON

Session LLL. Underwater Acoustics IX: Ambient Noise (Précis-Poster Session)

Alick C. Kibblewhite, Cochairman
Department of Physics
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Auckland, New Zealand

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Contributed Papers

Posters should be set up before 8:00 a.m. (before start of Session III). Following presentation of the précis, posters will be on display until 12:00 noon.

10:40

LLL1. Angular distribution of ice cracking events in the Arctic. P. Zakarauskas (Defense Research Establishment Pacific, FMO, Victoria, British Columbia V0S 1B0, Canada)

The measured vertical directionality of ice events in the polar ice pack is presented. Concurrent 345-s samples of ambient noise measured with four vertically spaced hydrophones suspended over the Arctic continental shelf were searched for corresponding peaks in the envelope of the acoustic pressure. In a previous analysis [P. Zakarauskas, J. Acoust. Soc. Am. Suppl. 1 **82**, S30 (1987)], the ice events were shown to be of thermal origin. The resulting angular distribution of the 765 ice events observed during the sample has its maximum at 15 deg below the horizontal, with most of the events arriving between 0 and 25 deg below the horizontal. An explanation for this observation is given in terms of a simple ray-based model of the propagation of ice cracking noise. The model includes the seafloor and under-ice angular reflection coefficients measured at a neighboring location.

10:44

LLL2. Abstract withdrawn.

10:48

LLL3. A standard definition for wind-generated, low-frequency ambient noise source levels. William M. Carey, William A. VonWinkle, David G. Browning (New London Laboratory, Naval Underwater Systems Center, New London, CT 06320), and Douglas J. Kewley (Maritime Systems Division, DSTO-WSRL, Salisbury, 5108, Australia)

Low-frequency, wind-generated ambient noise source levels are important input parameters for newly developed ambient noise prediction models such as DUNES. However, there has been a significant variation among recently reported source levels. An analysis is made of these values. Although the actual noise measurements, the assumed source model, and a small angle approximation are similar for all papers, differences arise due to additional geometrical factors or a further approximation to shift the source level to the surface. A standard source level definition and evaluation method are proposed. Standard values of low-frequency, wind-generated source levels are presented based on this strategy. [Work supported by NUSC.]

10:52

LLL4. A multipath calculation of surface-generated underwater acoustic ambient vertical directivity. Robert M. Kennedy and Thomas K. Szlyk (Naval Underwater Systems Center, AUTEC, West Palm Beach, FL 33402-7517)

A general calculation of the spatial correlation function due to a distributed acoustic source near the water surface has been made [W. A. Kuperman and F. Ingenito, J. Acoust. Soc. Am. **67**, 1988–1996 (1980)] using normal mode theory. In this paper, a multiple ray path analysis is used to calculate the acoustic ambient vertical directivity function. Multipath propagation and ambient directional functions are physically intuitive concepts to underwater acoustic theory and direct application of these notions has advantages in the interpretation of data. Existing procedures and computer code [H. Weinberg, J. Acoust. Soc. Am. **58**, 97–109 (1975)] were used to expand the received pressure field, from a distributed source, into a sum of terms interpreted as multiple propagation paths. Standard forms for the source function and a geometric transformation were used to convert the pressure field to a solid angle density function. Surface roughness, bottom geoacoustic parameters, and sound velocity–depth profiles measured in the Tongue of the Ocean were used in the calculations to predict hydrophone array performance in that area. Directivity function versus elevation angle and frequency were displayed for observers above and below a surface duct. A path-by-path contribution to the vertical distribution of energy is discussed.

10:56

LLL5. VLF ambient noise measurements in selected ocean environments. Hassan B. Ali and C. A. Fisher (Naval Ocean Research and Development Activity, John C. Stennis Space Center, MS 39529-5004)

As part of its ongoing program in very-low-frequency (VLF) acoustic propagation, the Naval Ocean Research and Development Activity has conducted experiments in several diverse geoacoustic environments. Using a vertical array of hydrophones in the water column and a number of ocean bottom seismometers on the seafloor, ambient noise measurements were made in both shallow (100 m) and deep (several km) water areas. The resulting seismoacoustic ambient noise data are analyzed in terms of their spatial and temporal characteristics, depth dependence, and frequency behavior. Spatial correlations of the hydrophone outputs as a function of depth are used to assess the spatial homogeneity of the noise fields. Not unexpectedly, the characteristics of the noise fields are markedly different in the various geographic areas. The differences are attributable in part to the different sources (shipping, wind, storms, etc.) and in part to the diverse bottom types involved. The potentially significant role played by propagation conditions in determining ambient noise levels is also discussed.

11:00

LLL6. The acoustic field of an oscillating bubble near a free surface. Hugh C. Pumphrey and L. A. Crum (National Center for Physical Acoustics, P.O. Box 847, University, MS 38677)

Bubbles may be entrained into the ocean by a variety of natural processes such as rainfall and breaking waves. It is often postulated that the free oscillations of these bubbles at their resonance frequencies are a significant source of ambient noise in the ocean. Experimental results will be presented which show that a bubble entrained near the surface has the resonance frequency and damping constant predicted by theory, with deviations that may be explained by such factors as the nonsphericity of the bubble and the proximity of the free surface. The bubble is a simple source within a small fraction of a wavelength of a free surface and should therefore have a dipole radiation pattern. Measurements of the field were made; these show that the radiation pattern is as predicted and allow calculation of the amplitude of oscillation of the bubble. It is known from previous work that raindrops in a certain size range will entrain bubbles. Using the known source strength of a bubble, the spectrum level produced by a typical rain shower is calculated and compared with measurements made of real rain. [Work supported by ONR.]

11:04

LLL7. Offshore measurements of the underwater acoustic noise spectrum during rainfall. Paul Kraeutner (JASCO Research Limited, 9865 West Saanich Road, Sidney, British Columbia V8L 3S1, Canada)

The underwater acoustic noise spectrum during rainfall has been measured in a coastal ocean environment for a wide range of sea states and rainfall intensities. The relationship between spectral noise level and rain rate was examined and a power law function fitted to data at 5, 8, 15, 20, and 30 kHz. Finally, a family of curves representing the acoustic noise spectrum for sea states $\frac{1}{2}$, 1, 2, 3, 4, and 5 and rain conditions of 1.0, 3.0, 5.0, and 7.0 mm/h was derived. Results indicate a domination of the acoustic noise spectra by rain-generating mechanisms at frequencies in the band 7–25 kHz and sea states less than 4. Furthermore, the smearing of the rain-generated spectral peak near 15 kHz strongly supports the postulation of a strong dependence of rain-generated acoustic noise on sea surface conditions. [Work supported by DSS Unsolicited Proposals Program, IOS, and DND.]

11:08

LLL8. The hydroacoustics of a drop impact. Jeffrey A. Nystuen (Code 68Ny, Naval Postgraduate School, Department of Oceanography, Monterey, CA 93943)

Rainfall has consistently been one of the most difficult geophysical quantities to measure, especially in oceanic regions. Recently, it has been demonstrated that rainfall can be detected at sea by monitoring for the unique underwater sound generated by the rain impacting onto the ocean surface [Nystuen and Farmer, Atmos-Ocean (1988), in press]. The goal of quantifying rainfall rate through underwater ambient sound measurements requires a more complete understanding of the hydroacoustics of the drop splash and the influence of wind upon these physics. A numerical model is used to explain the hydroacoustics of a drop impact as an acoustic waterhammer modified by the emergence of a fluid jet at the base of the drop. The drop is an acoustic source until the radius of the contact circle between the drop and the underlying water surface reaches the maximum cross-sectional radius of the drop. The duration of the pulse depends on drop size, shape, and speed. Using realistic raindrops, the observed unique underwater acoustical signature of rain, an increase of ambient sound energy at 15 kHz, is explained.

11:12

LLL9. Radiated noise increase of a submerged turbulent jet flow due to bubble entrainment. M. S. Korman (Physics Department, U.S. Naval Academy, Annapolis, MD 21402 and National Center for Physical Acoustics, University, MS 38677), H. C. Pumphrey, and L. A. Crum (National Center for Physical Acoustics, University, MS 38677)

Experiments are performed in an open tank ($L = 1$ m, $W = 0.75$ m, $H = 0.6$ m) to study the effects of entrained air bubbles on the noise radiation produced by a turbulent shear flow created by a submerged water jet (nozzle diameter $D = 4$ mm located 27 cm from the bottom). Using a small hydrophone, measurements of the average frequency spectrum level between 1.9 and 20 kHz are compared for case (1) no bubble entrainment and case (2) bubble entrainment, under the same experimental conditions of the turbulence. A 2.54-cm column of bubbles (with radii between $\frac{1}{4}$ mm and $1\frac{1}{2}$ mm) is generated below the nozzle exit that becomes entrained in the jet core between 4 and $8D$ from the exit. Nozzle exit velocities U are chosen below the cavitation threshold of the jet [$\sigma = 2(P_{\rm amb} - P_{\rm vap})/\rho U^2$ and $\sigma > 0.4$]. Results indicate an increase of over 10 dB in the radiated noise level measured between 6 and 10 kHz and over 20 dB between 10 and 20 kHz for studies where $0.42 < \sigma < 1.47$. It is hypothesized that bubble breakup may be one cause of this increase. [Work supported in part by the ONR.]

11:16

LLL10. Remote sensing using hydroacoustics. Paul H. Patrick, B. Sim, and G. Hunt (Ontario Hydro Research Division, 800 Kipling Avenue, Toronto, Ontario M8Z 5S4, Canada)

Laboratory experiments involving hydroacoustics (sonar) were conducted in a semi-anechoic tank (15.5 m long × 15.5 m wide × 1.5 m deep) as a remote sensing technique to collect quantitative information on fish. Initial research was directed on the capability and accuracy of commercial sonar equipment in identifying various fish species and geometric shapes. Because of bandwidth limitations in the receiver, this equipment was limited in the ability to identify targets especially at the lower pulse widths (0.1 ms). Modifications to this equipment (wideband detector module) resulted in greater spatial resolution of targets (7.5 cm), and more detailed information in the various backscattering curves required for species identification.

11:20

LLL11. Device for reducing noise in underwater acoustics. J. M. Wagner (ECAN St. Tropez, Section Recherche, 83990 St. Tropez, France)

In the case of an underwater acoustics antenna, the transducer is usually mounted as follows: One face of the sensor is leaned against a structure withstanding the pressure, and the other one is in contact with water. Any structure vibration is transmitted to the sensor and generates noise that alters the receiving performances. It will be shown that, in the case of an asymmetrical mounted transducer, the sensor sensitivities depend on the excitation direction. This feature is applied in inserting a pressure sensor between the transducer support and the structure. The response of this system is analyzed and shows that the S/N ratio can be significantly increased by subtracting noise. Trials in an acoustic water tank are described and results on real antenna are given.

Session MMM. Musical Acoustics VI: Musical Scales and Tuning in Eastern and Western Music

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Tomoyasu Taguti, Cochairman
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Chairman' Introduction—2:00

Invited Papers

2:05

MMM1. The cognitive framework for musical scales in various cultures. W. Jay Dowling (Program in Human Development and Communication Sciences, UT/Dallas, Richardson, TX 75083-0688)

Musical scales in different cultures throughout the world share certain design features. This suggests that some general constraints influence scale structure. Some features of scales reflect constraints apparently grounded in the physical stimulus and perhaps the auditory system; for example, octave equivalence. Other features appear to derive from limitations on human information processing abilities; for example, the use of a limited number of discrete pitch levels. The latter seems to arise from the advantages of discrete categorization of elements in noisy communication channels, and cognitive limitations on the number of categories humans can handle at a time. Combining these two types of constraints further specifies the form that scales can take; octave equivalence combined with a limited number of pitch categories leads to scales having up to seven or so pitches per octave, with the pitch categories recycled through several octaves in a logarithmic pattern. Subtler constraints are discussed, and the resulting design features are illustrated with examples from several cultures.

2:35

MMM2. Absolute and relative pitch identification by absolute pitch possessors. Ken'ichi Miyazaki (Department of Psychology, College of General Education, Niigata University, Niigata, 950-21 Japan)

The identification of absolute and relative pitch was investigated with absolute pitch (AP) possessors and nonpossessors as subjects. In an AP experiment, isolated tones of different pitches (including microtonal pitches) were presented as stimuli in random order. Subjects were requested to select and press a key corresponding to each stimulus on a musical keyboard. While the responses of non-AP subjects were widely dispersed, the AP subjects could categorize the presented tones quite consistently. Specifically, they identified white-key notes more correctly and rapidly than black-key notes. In a relative pitch experiment, isolated melodic ascending intervals from the minor second to the octave were randomly presented. The lower tone of an interval was either middle C, F♯, or a 50-cent higher D♯. Then non-AP subjects performed more or less successfully and almost equally regardless of transposition. On the other hand, the AP subjects made more errors and gave longer response times when the lower tone of an interval was F♯ or the higher D♯ than when it was C. These results suggest that at least some AP possessors have difficulty in dealing with musical intervals and tend to rely upon their AP sense in the face of a relative pitch task.

3:05

MMM3. Two approaches to tuning and pitch systems. Gerald J. Balzano (Department of Music, University of California at San Diego, La Jolla, CA 92093)

This paper attempts to sort out some distinctions important to the understanding of tuning and pitch systems. Two different approaches are distinguished, the *acoustic* approach and the *structural* approach. In the acoustic approach, intervals are defined as whole-number ratios, and a pitch system considered as a collection of ratios, some interrelated, some incommensurable. On the structural approach, the system is taken as prior to the intervals that compose it; intervals, then, become elements in a system with properties at least partially determined by the system. In the acoustic approach, what the intervals *are* and how they are *tuned* amount to the same thing, so it makes no sense to distinguish between a *pitch system* and a *tuning scheme*. But the structural approach distinguishes sharply between the pitch system as an abstract template or structure, and a tuning scheme as a strategy for mapping the template to specific frequency relations. Acoustic and structural views of tonality and microtonality will be presented, including examples of pitch systems constructed on both approaches. The present terminology will be applied to diverse instances of musical thought and practice, and implications of the two approaches for future evolution of pitch systems will be discussed.

3:35

MMM4. Modeling the acquisition of cross-cultural differences in tonality with neural nets. Jamshed J. Bharucha (Department of Psychology, Dartmouth College, Hanover, NH 03755)

The acquisition of scale schemas through cultural exposure was modeled with neural net simulations. Associative networks employing the back-propagation learning rule were exposed repeatedly to scale sets of one culture and were then tested for their ability to complete scale subsets of either the same culture or a different culture. The scales were taken from Western and North Indian music. Each network completed the scale subsets so as to most closely match the scales of its own culture. When the foreign scale subset was sufficiently different from any scale to which the network had been exposed, the network showed no clear preference for scale completion. In some cases, the foreign scale was interpreted as being similar to a native scale with a shifted tonic. It is argued that acoustical patterns that occur with considerable regularity in the environment become encoded through passive exposure and serve to generate expectations in context. Reaction time tasks were employed to study these expectancies empirically, for both Western and Indian music.

4:05

MMM5. Koto scales and tunings. Masateru Andô (1-3-17 Sakuragaoka, Setagaya-ku, Tokyo, 156 Japan)

The koto, an instrument consisting of 13 strings stretched over a long sounding board, is tuned by a set of movable bridges, one for each string. The koto's musical range normally extends from A^1 to A^7. Within this range, two and a half octaves are most commonly utilized. Excluding modern pieces, most koto pieces are constructed from a five-tone scale. Generally classified, there are two types of scales that can begin on any tone; a minor-sounding scale (mi-fa-la-si-do-mi), and a major sounding scale (re-mi-sol-la-si-re). The most common tuning is known as *Hira-jôshi*, which begins on the fifth string of the koto and follows the minor-sounding scale noted above. Tuning on the koto is accomplished by first setting string No. 1 to the desired pitch. Then the fifths, fourths, octaves, firsts, and minor seconds are tuned, creating a scale close to the Pythagorean scale. Like piano tuning, inharmonicity is observed. Several of the traditional tunings for koto music have the same scale as the minor-sounding *Hira-jôshi* except that they begin on a different string; No. 6, 7, 8, or 9, rather than string No. 5. Likewise, the major-sounding scales can begin on any of the same five strings.

FRIDAY AFTERNOON, 18 NOVEMBER 1988 HONOLULU/KAHUKU ROOM, 2:00 TO 3:11 P.M.

Session NNN. Noise IV and Architectural Acoustics IX: Vehicle Interior Sound and Vibration

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Chairman's Introduction—2:00

Invited Papers

2:05

NNN1. Sound intensity measurements in a vehicle's interior sound field. Ken'ichiro Suzuki and Tetsuo Maki (Vehicle Research Laboratory, Central Engineering Laboratories, Nissan Motor Company, Ltd., Natsushima-cho, Yokosuka, 237 Japan)

Sound pressure and sound intensity in the passenger compartment of a vehicle with many standing waves are measured through piston–plate excitation. These measurements provide the characteristics of attenuation

in the free field near the sound source, which is found to be approximately -6 dB per doubling of distance. At a certain distance from the sound source, sound intensity shows an average sound-pressure level between the maximum and minimum values in the passenger compartment. The reason for this is that, owing to the use of sound-absorbing materials, the sound field does not consist only of standing waves. It is revealed that progressive waves dominate near the sound source. As a result, sound intensity measurements in passenger compartments provides the precise location and contribution of the sound source, even when booming sounds are present. Moreover, using the modal analysis method to investigate the behavior of the body vibration from the enforced point to the sound source is effective in improving the body structure for reducing passenger compartment noise levels.

2:20

NNN2. A method for evaluating the interior sound quality of automobiles. Tsuyoshi Yamashita, Hajime Niikura, Mitsuo Nakamura (Honda Research and Development Tochigi Center, Shimotakanezawa, Haga, Tochigi, 321-33 Japan), and Otoichi Kitamura (Osaka University of Arts, Higashiyama, Kanan-cho, Minamikawachi, Osaka, 585 Japan)

The A-weighted sound-pressure level has been the conventional method of evaluating interior car noise aimed at reducing loudness. However, it is difficult to evaluate the quality of sound only by loudness. The multivariate analysis method was applied using data obtained from a semantic differential questionnaire with over 20 items about four different car models' interior sounds. The results showed that the bias from the image of the models had a larger effect on impressions than the actual sound. This shows that a blindfolded test is very important. Thus, the test procedure was modified to obtain correct results, and a blindfold test was adopted. The test was done with 48 subjects monitoring both the actual sound of the automobiles and their recorded sound. The data were analyzed using the principal component analysis, and three main factors in the sound were revealed: "beautifulness," "powerfulness," and "metallicness." These factors are similar to those already found in studies of other types of sound.

Contributed Papers

2:35

NNN3. Sound reproduction in a car cabin using a digital filter network. Michio Hanba, Hareo Hamada, Tanetoshi Miura (Tokyo Denki University, 2-2 Kanda-Nishiki-cho, Chiyoda-ku, Tokyo, 101 Japan), and Yoshinori Kiryu (Automobile Audio Visual Equipment Factory, Hitachi, Ltd., 1410 Inada, Katsuta, 312 Japan)

Designs to reproduce stereo sound fields in a car cabin have many problems, such as the asymmetrical positions of loudspeakers (for the listener) and the complex characteristics of the sound transmission. Therefore, a technique is investigated in which the sound field in a car cabin is transformed using a digital filter network, keeping in mind the interaural sound pressure in both ears of the listener. The digital filter network can be realized from the transfer functions measured by a head and torso simulator in the car cabin and in the desired sound field. The purpose of using the digital filter network is to equalize the sound pressure in both ears of the listener (when the digital network is not used) to that obtained in the desired sound field. The realizability of the desired sound field in a car cabin was confirmed by computer simulation.

2:47

NNN4. A study of noise reduction of mufflers. Tsuyoshi Nishimura (Kumamoto Institute of Technology, 4-22-1 Ikeda, Kumamoto, 860 Japan), Masanao Ebata (Faculty of Engineering, Kumamoto University, 2-39-1 Kurokami, Kumamoto, 860 Japan), and Josuke Okda (Faculty of Engineering, Kyushu Tokai University, 223 Oh'e-Toroku, Kumamoto, 862 Japan)

There are two methods of analyzing the noise reduction characteristics of mufflers. One is a method based on acoustic theory. This method is widely used, because the computation is relatively simple and the noise reduction in each component of the muffler can be easily evaluated from the solution. However, the computation is carried out under assumptions that the system is linear, isentropic, and so on, and the obtained results do not agree with those obtained in practice. Another is the method of characteristics that gives a more accurate solution although the calculation is complicated. Thus the method becomes effective and convenient for practical application if the results obtained by the acoustic theory can be corrected based on the strict solutions obtained by the method of characteristics. This report deals with the method of correction for various shapes of mufflers composed of a straight pipe and an expansion chamber.

2:59

NNN5. Cockpit noise generation mechanisms in light aircraft. Fred C. De Metz, Sr. (17145 San Fernando Mission Boulevard, Granada Hills, CA 91344)

The high noise levels occurring in the cockpits of light, powered aircraft can significantly contribute to pilot fatigue, interfere with pilot communications with passengers and air traffic control, cause permanent hearing loss, and detract from an otherwise pleasurable flight experience. The levels and spectral features of the cockpit noise field are presented for a number of popular types of light aircraft, under different operating conditions. The dominant noise generation mechanisms are identified and the effectiveness of practical and impractical noise reduction techniques for existing light aircraft are estimated. The noise control design features for a new generation of light aircraft are explored.

Session OOO. Physical Acoustics IX: Anisotropy, Scattering, Propagation, Diffraction, and Refraction

Gilles A. Daigle, Cochairman
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Contributed Papers

2:00

OOO1. Characterization of anisotropic elastic wave behavior in media with parallel fractures and aligned cracks. Chaur-Jian Hsu and Michael Schoenberg (Schlumberger-Doll Research, Old Quarry Road, Ridgefield, CT 06877-4108)

Ultrasonic experiments were performed to investigate the anisotropic properties of an isotropic background medium with a system of long closely spaced parallel fractures. Immersion experiments at frequencies from 0.1–0.7 MHz were performed on Plexiglas that was cut into thin sheets and then pressed together. Measurements of phase and group velocity were obtained for compressional and shear waves both normal to and parallel to the "fractures" and for quasicompressional and quasishear waves at a variety of oblique angles. The data were analyzed to determine the anisotropy parameters due to the fracturing, based on modeling the fractures as a set of infinite, linear, although not necessarily rotationally symmetric, slip interfaces [M. Schoenberg, J. Acoust. Soc. Am. Suppl. 1 **82**, S88 (1987)]. For the most part, the experiments fit the theory quite well even though losses that can occur in the fractures have been ignored in this first attempt to match theory and experiment. The anisotropy attributable to the fracture system depends on the geometry of the fractures, the roughness, the static closing stress across the fractures, and the nature of the infilling medium, if any. From a knowledge of the anisotropic elastic moduli of the fractured medium and the isotropic background moduli, the average compliance matrix may be deduced. In turn, this gives the average behavior of each fracture plane.

2:12

OOO2. Decoupling fracture parameters from background transverse isotropy in orthorhombically symmetric media. Julie A. Hood (Hawaii Institute of Geophysics, 2525 Correa Road, Honolulu, HI 96822) and Michael Schoenberg (Schlumberger-Doll Research, Old Quarry Road, Ridgefield, CT 06877-4108)

A system of plane parallel fractures or periodic fine layering renders any elastic medium seismically anisotropic. The anisotropy increases in complexity as the number of different systems incorporated into the medium increases. Using the group calculus formulation for layered media developed by Schoenberg and Muir [submitted to Geophysics], the effects of each individual system can be separated arithmetically after the properties of each system in the anisotropic medium are transformed to an element of an Abelian group. In an orthorhombic medium resulting from fractures orthogonally embedded in a transversely isotropic (TI) background, the contribution of the fractures to the stiffness matrix can readily be subtracted out, reducing the number of moduli from eight to five. The removed moduli provide information on the fracture properties, such as average fracture spacing, microcrack density, and average compliance of fracture-filling material. The remaining simplified (five-parameter) stiffness matrix can be analyzed further to infer the characteristics of the background TI medium. [Work supported by NSF.]

2:24

OOO3. Computer synthesis of acoustic wave fields in laminated and fractured media. Subhashis Mallick and L. Neil Frazer (HIG, University of Hawaii at Manoa, Honolulu, HI 96822)

Frequency-wavenumber integration is used to compute time-domain sonic waveforms at any point in a stratified medium. The source is either a point force, or a point dislocation, at any depth. The medium consists of welded homogeneous layers with all 21 elastic constants varying independently from layer to layer. Currently, a free-surface boundary condition at the top of the medium and a radiation condition at the bottom is used, but these boundary conditions can be easily changed. Although final results are in the time domain, the basic calculations are carried out in the frequency domain so anelastic effects are incorporated by use of complex elastic constants. (Usually, instead of directly specifying the elastic constants in each layer, Schoenberg algorithms were used to "fracture" a layer or to "construct" a layer from many thin microlayers of simpler material. These algorithms have a group property [Schoenberg and Muir, 1988] which makes it easy to generate fractured laminates.) For elastic materials with azimuthal anisotropy, two wavenumber integrations are required. If the medium is many wavelengths thick, and the source and receiver(s) are many wavelengths apart, then several hours of Cray CPU time are needed to generate a complete response. [Work supported by Geo-Pacific Corporation.]

2:36

OOO4. Recent applications of acoustoelasticity. P. P. Delsanto, P. Quarati (Dip. di Fisica del Politecnico, Torino, Italy), F. Boschetti (Aeritalia GVC, Corso Marche n. 41, Torino, Italy), H. H. Chaskelis, and R. B. Mignogna (Naval Research Laboratory, Code 6385, Washington, DC 20375)

A perturbation treatment of the propagation of Rayleigh waves on the surface of a homogeneous initially deformed material plate has been recently proposed [P. P. Delsanto and A. V. Clark, J. Acoust. Soc. Am. **81**, 952–960 (1987)]. Although an "exact" (numerical) solution is also possible, the perturbation approach has the distinctive advantage of allowing the solution of the "inverse problem" of determining texture and/or stress from the results of Rayleigh waves time-of-flight measurements at different propagation angles. Based on this perturbation approach, nondestructive techniques for the evaluation of the elastic constants, texture, and stress [Delsanto *et al.*, submitted to J. Acoust. Soc. Am.] have been proposed for the special case of polycrystalline aggregates, which can be assumed to consist of a slightly orthotropic distribution of cubic crystallites. Typical examples are aluminum alloys. An analysis of the experimental results and a comparison with other NDE techniques confirm the reliability of this method.

2:48

OOO5. Scholte–Gogoladze-like waves at the interface between a fluid and an anisotropic layered composite. Arthur M. B. Braga and George Herrmann (Division of Applied Mechanics, Stanford University, Stanford, CA 94305-4040)

It is now well understood that two types of plane harmonic waves can propagate freely along the plane interface between a homogeneous isotropic elastic solid and an acoustic fluid. The first, with decaying amplitude as it propagates along the interface, and which is equivalent to the Rayleigh wave in the solid as the fluid density becomes negligible, is called a *leaky* or *generalized Rayleigh* wave. The other, named here the *Scholte–Gogoladze* wave after the two researchers who first, independently, uncovered its existence in 1948, propagates unattenuated parallel to the boundary, while decaying exponentially in both directions away from the interface. In the present contribution, the existence of Scholte–Gogoladze-like waves at a fluid/layered composite interface is investigated. The composite is made-up of an arbitrary number of homogeneous anisotropic layers stacked in a unit cell that repeats periodically. An extension of Stroh's sextic matrix formalism to periodically layered media is employed to carry out the analysis. The associated eigenvalue problem is solved in closed form, and the frequency equation for the interface waves is formulated in terms of the *surface impedance tensors* for both the fluid and the layered composite. In contrast to the homogeneous medium, the Scholte–Gogoladze-like waves are now dispersive and, for certain combinations of the material parameters, the dispersion spectrum consists of more than one branch in the frequency-wavenumber space. [Work supported by ONR.]

3:00

OOO6. Inhomogeneous plane waves: Energy conservation and reflection coefficient. O. Leroy, J. M. Claeys (K. U. Leuven Campus Kortrijk, B-8500 Kortrijk, Belgium), and G. Quentin (Université Paris 7, Groupe de physique des solides de l'Ecole Normale Supérieure, Paris, France)

In studying the reflection and transmission of inhomogeneous plane waves at liquid–solid interfaces, it is found that the theory predicts a minimum in the reflection coefficient at the Rayleigh angle. This phenomenon was not predicted previously using homogeneous plane waves. Even surprising is the fact that the modulus of the reflection coefficient of inhomogeneous plane waves becomes greater than unity for angles larger than the Rayleigh angle. It is shown that this phenomenon is consistent with the principle of conservation of energy and a physical interpretation leads to explanation of the shift of those waves along the surface (Schoch displacement).

3:12

OOO7. Exterior acoustic scattering/radiation without interior eigenvalue problems. Richard Paul Shaw (Department of Civil Engineering, SUNY at Buffalo, Buffalo, NY 14260)

The use of integral equation formulations, such as boundary-element methods or *T*-matrix methods, for the solution of time harmonic exterior acoustic radiation or scattering is hindered by the presence of the "fictitious interior eigenvalue" difficulty that causes the basic integral equation formulation, and thus any numerical scheme based on such a formulation, to fail at certain frequencies related to the interior acoustic problem. A number of techniques have been developed in the last 20 years to circumvent this difficulty; these involve additional computational effort. A method is presented here that eliminates this difficulty analytically by embedding the original obstacle in an infinite acoustic fluid bounded by a simple surface of sources and/or doublets. Such an approach moves the eigenvalue failures from those for the original shape, which are in general unknown, to those for a simple shape which are known. These eigenvalues can be carried through the formulation analytically and canceled exactly, thereby eliminating any subsequent numerical failures.

3:24

OOO8. Dispersion curves for thermomechanical waves. Anthony J. Rudgers (Naval Research Laboratory, Underwater Sound Reference Detachment, P.O. Box 568337, Orlando, FL 32856-8337)

When mechanical deformation and heat conduction are both accounted for, two coupled partial differential equations (PDEs) describe wave propagation in a medium. For one-dimensional wave propagation in a medium without mechanical dissipation, these PDEs yield a dispersion relation of complex quadratic form, which shows that two kinds of waves can propagate. Each wave has a mechanical and a thermal component. The speeds and the attenuations of such waves are illustrated in two cases. First, it is shown that the "classical" form of the thermomechanical PDEs provides an interesting but nonphysical description of wave propagation. Second, a Vernotte thermal relaxation time is introduced into the PDEs. When the value of the Vernotte parameter is set by applying the Debye theory of specific heats in a rudimentary way, it is shown that a physically plausible description of wave propagation can be obtained. In this latter connection, the consequences of using a "phonon-gas" model of a thermomechanical medium is considered briefly. [Work supported by ONR.]

3:36

OOO9. Measurement of the temperature dependence of the piezo-optic coefficients of liquids using the submersible electrostatic acoustic transducer. William T. Yost and John H. Cantrell (NASA-Langley Research Center, Hampton, VA 23665-5225)

The development of a liquid-immersible megahertz-range electrostatic acoustic transducer was previously reported. The transducer consists of a thin conductive membrane stretched over an electrically isolated cylindrical electrode recessed approximately 10 μ from the membrane inner surface. The transducer is shown to be capable of measuring absolute ultrasonic displacement amplitudes in the subangstrom range. This capability, combined with Raman–Nath light diffraction measurements, permits the measurement of the piezo-optic coefficients of liquids. Measurements of the piezo-optic coefficient in distilled water as a function of temperature are reported. The results are compared with values obtained from other techniques.

3:48

OOO10. Forward-glory scattering from a spherical shell and backscattering from a convex hemispherical shell. Steven G. Kargl and Philip L. Marston (Department of Physics, Washington State University, Pullman, WA 99164)

Observations were made of the scattering of short tone bursts from hemispherical and empty spherical stainless-steel shells in water. Lamb-wave contributions to the backscattering from spherical shells have been previously described [S. G. Kargl and P. L. Marston, J. Acoust. Soc. Am. Suppl. 1 **81**, S14 (1987)] modeled by an elastic generalization of GTD [P. L. Marston, J. Acoust. Soc. Am. **83**, 25–37 (1988)]. In the present research, novel focal or "glory" properties of Lamb-wave contributions to forward scattering from the sphere and to backscattering from the hemisphere are described. The observations of a strong forward glory are important since it follows from the *optical theorem* that Lamb waves will strongly affect the *total* cross section. The observed focal properties show that the scattered waves appear to emanate from virtual ringlike sources describable from geometrical considerations. The forward-glory scattering arrived earlier than the ordinary forward diffraction; here, the Lamb waves have phase velocities $c_l > c$ for water. Typically $ka \approx 57$, where the sphere's outer radius $a = 19$ mm and thickness $= 3$ mm. The backward glory from the hemisphere occurs as a consequence of single and multiple reflections of Lamb waves from the unconstrained edges of the hemisphere; the radius b_l of the virtual source is $\approx ac/c_l$. These echoes were preceded by a specular echo. [Work supported by ONR.]

4:00

OOO11. Calculation of sound diffraction by means of the integral equation. Yoshihiro Furue (Department of Architectural Engineering, Kyoto University, Sakyo-ku, Kyoto, 606 Japan)

The Helmholtz integral equation and its normal derivative form have been widely used to calculate the acoustic diffraction by any shaped ob-

jects. A method for calculating the matrix of the simultaneous algebraic equations derived from the above integral equations is proposed to save computation. All of the four kinds of surface integral should be calculated to determine the elements of the matrix and converted to the marginal integral of the tangential surface elements of the object. The velocity potential, or its normal derivative on each surface element is assumed to be constant and the calculation point is located at its center. Numerical examples for the sound diffraction by a rigid thin pipe of finite length by means of the above method are presented with measurements. In this case, the nonderivative Helmholtz integral equation cannot be used because of degeneration. Only the derivative form is applicable, in which the unknowns should be the difference in the velocity potential between the front surface and the back.

4:12

OOO12. Application of the integral equation method to two-dimensional sound diffraction problems. Yoshinari Horinouchi and Yoshihiro Furue (Department of Architectural Engineering, Kyoto University, Sakyo-ku, Kyoto, 606 Japan)

This report is concerned with the numerical calculation of two-dimensional sound diffraction fields around thin rigid plates by the integral equation method. Generally, the basic equation used in this method is the Helmholtz integral equation. However, if the objects investigated are infinitely thin plates, the basic equation cannot be applied. In such cases, the normal derivative form of the Helmholtz integral formula (NDF for short) is useful [T. Terai, J. Sound Vib. 69(1), 71–100 (1980)]. When the plate is flat and rigid, the equation to be solved is reduced to an integral equation of the first kind by means of NDF. And, as the fundamental solution in two dimensions is not $\exp(ikr)/r$, but is a zero-order Hankel function of the first kind, the treatment of singularities is more complicated. The numerical examples treated in this report are problems of an acoustic barrier consisting of several rigid and flat plates. It is shown that the numerical solutions agree with the analytical ones and also with the experimental results.

4:24

OOO13. Experimental simulation of a pressure-release finite cylinder using a closed-cell neoprene coating. Gary W. Caille, Peter H. Rogers, and Jacek Jarzynski (George W. Woodruff School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

The scattered field of a coated finite cylinder with flat end caps is compared with the theoretically obtained field for a pressure-release cylinder. The cylinder is stainless steel with a radius of 2.0 in. and length of 12.75 in. The wall thickness is 0.25 in. and the end cap thickness is 0.75 in. The ka range is approximately 1.5–15. The coating is 0.25 in. of closed-cell neoprene rubber. The backscattered pressure is calculated for sound incident normal to the axis using a modified Naval Oceans Systems Center CHIEF program and for end-on incidence using the Naval Research Laboratory SHIP program. The intent of the coating is to simulate the pressure-release boundary condition. This condition is, in fact, achieved because: (1) Agreement between the theoretical and experimental results is within 1 dB; (2) the coating removes various elastic effects including a return signal from the back end (observed for the uncoated cylinder, for the end-on case); and (3) the scattered signal from the coated target is phase inverted relative to the uncoated case. [Work supported by Office of Naval Research, Code 11250A.]

4:36

OOO14. Impulse sound propagation above a curved surface. G. A. Daigle (Division of Physics, National Research Council, Ottawa, Ontario K1A 0R6, Canada) and Richard Raspet (Department of Physics and Astronomy, The University of Mississippi, University, MS 38677)

Theory for the propagation of sound in an upward refracting atmosphere or, by analogy, above a curved surface, includes a residue series solution. Controlled experiments made indoors above a carefully constructed curved surface using pure tones have shown that the diffraction of sound energy deep within the shadow can be accurately calculated from the residue series solution [A. Berry and G. A. Daigle, J. Acoust. Soc. Am. Suppl. 1 **81**, S97 (1987)]. However, when the theory is used to predict impulse sound propagation in a refractive atmosphere, serious discrepancies are observed [C. G. Don and A. J. Crammond, J. Acoust. Soc. Am. **80**, 302–305 (1986)]. The calculated waveforms show considerably more attenuation and dispersion than the measurements. Recently, a more accurate calculation of the residue series has improved the prediction, but the discrepancy is still apparent [R. Raspet and S. J. Franke, J. Acoust. Soc. Am. **83**, 1964–1967 (1988)]. In order to elucidate the discrepancy, impulse sound measurements made indoors under controlled conditions above a curved surface will be described. The measurements will be compared with the predicted waveforms. Measurements are also planned outdoors under realistic conditions.

4:48

OOO15. Propagation of energy in nonspreading wave packets. Richard W. Ziolkowski, D. Kent Lewis (Lawrence Livermore National Laboratory, Livermore, CA 94550), and Bill D. Cook (Cullen College of Engineering, University of Houston, Houston, TX 77204-4792)

Is there anything new about the wave equation? Some investigators have found new, unusual solutions. Brittingham started it all in 1984 by publishing solutions describing localized packets of energy that propagated without spreading. A controversy arose as Brittingham's solutions had questionable features including infinite energy, a backward wave component, and a questionable launchability. Ziolkowski demonstrated that by combining many modes of Brittingham's solutions, these problems disappeared. This paper describes a brief history, the basic concepts, the objections, and the major breakthroughs to date. Experimental evidence suggests that a new solution can be launched. [Work performed under the auspices of the U.S. Dept. of Energy by Lawrence Livermore National Laboratory under Contract W-7405-Eng-48.]

5:00

OOO16. An experimental reach for the localized wave packet. Richard W. Ziolkowski, D. Kent Lewis (Lawrence Livermore National Laboratory, Livermore, CA 94550), and Bill D. Cook (Cullen College of Engineering, University of Houston, Houston, TX 77204-4792)

Ziolkowski's novel solutions to the wave equation (see previous abstract) suggest that an approximation form of these wave packets can be launched from a finite size array. An experiment was designed based on ultrasonic pulses whose spectra centers around 1 MHz. At these frequencies, and with proper choice of array shape and size, the novel features of these new solutions should be observable at distances less than 1 m from the array. The one-dimensional array had many unique features. It was synthetic, consisting of only one commercial NDE ultrasonic transducer and a laser beam–photodiode detector. By reversing the role of transmitter and receiver, the problem could be modeled as a line transmitter, and a point receiver. [Work performed under the auspices of the U.S. Dept. of Energy by Lawrence Livermore National Laboratory under Contract W-7405-Eng-48.]

5:12

OOO17. Calculation of turbulence effects in an upward refracting atmosphere. Kenneth E. Gilbert (National Center for Physical Acoustics, University, MS 38677) and Richard Raspet (Physical Acoustics Research Laboratory, University of Mississippi, University, MS 38677)

In an upward refracting atmosphere, measured values of excess attenuation (50–500 Hz) seldom exceed 20 to 30 dB at a range of 1 km. Calculations of excess attenuation for a steady (nonturbulent) atmosphere predict a deep shadow zone with much higher excess attenuation. This paper investigates the contribution of atmospheric turbulence to decreasing the predicted excess attenuation. Since no convenient analytical method presently exists which can simultaneously account for turbulence, upward refraction, and a finite-impedance ground surface, a parabolic equation method is used to numerically simulate sound propagation. As an initial test, the calculation is compared to Daigle's theoretical and experimental results for a homogeneous atmosphere [J. Acoust. Soc. Am. **65**, 45–49 (1979)]. For a test with upward refraction, the calculation is compared to the experimental results of Wiener and Keast [J. Acoust. Soc. Am. **31**, 724–733 (1959)].

5:24

OOO18. Free-field nonlinear pulse propagation outdoors. Victor W. Sparrow (Department of Electrical and Computer Engineering, University of Illinois, Urbana-Champaign, and USA-CERL, P.O. Box 4005, Champaign, IL 61820-1305) and Richard Raspet (Physical Acoustics Research Group, The University of Mississippi, University, MS 38677)

As an intermediate step toward developing a two-dimensional finite difference model of outdoor sound pulse propagation containing both cumulative finite amplitude effects and the local effect of a finite impedance ground, a program addressing the nonlinear steepening has been implemented. Previous studies [J. Acoust. Soc. Am. Suppl. 1 **80**, S104 (1986) and J. Acoust. Soc. Am. Suppl. 1 **82**, S64 (1987)] have involved linear propagation only. The vectorized code has been running on a CRAY X-MP at the National Center for Supercomputer Applications in Champaign, IL because of its intensive CPU and memory requirements. Details of the time stepping algorithm, as well as those of the numerical boundary conditions, will be given. The results of this program will be compared with that of SHOCK.F, a one-dimensional acoustic propagation code including finite amplitude effects [J. Acoust. Soc. Am. **74**, 1514–1517 (1983)]. As an example of some typical color graphics output, a short videotape will conclude the presentation.

5:36

OOO19. A model of sound propagation in a temperature inversion near the ground. W. K. Van Moorhem (Department of Mechanical and Industrial Engineering, University of Utah, Salt Lake City, UT 84105)

A model has been developed for three-dimensional sound propagation from a point source in a realistic vertical sound-speed profile corresponding to a temperature inversion near the ground surface. The ground itself is modeled as a finite normal impedance surface. The model is developed by Hankel transforming the three-dimensional wave equation with a variable vertical sound speed. An approximate analytic solution of the transformed equation is then obtained, valid for wavelengths small compared to the scale of the temperature variation. The form of this approximate solution depends on whether the source is above or below the receiver and whether rays can pass directly from the source to receiver, are reflected from the ground or are refracted by the temperature gradient. The Hankel transformed solution is returned to physical space by a numerically calculated inverse transformation. The solutions obtained show the expected interference pattern and sound-pressure level decay near the source, but no significant change in the rate of decay is observed from that of the isothermal atmosphere. [This work was supported by the NASA Langley Research Center under Grant NAG-1-283.]

5:48

OOO20. Experimental validation of the MAE theory of sound propagation over two consecutive ridges of finite impedance. James A. Kearns, Yves H. Berthelot, and Ji-xun Zhou (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

A difficult problem within the realm of outdoor sound propagation is predicting the acoustic field associated with long-range sound propagation over hilly terrain. Such terrain is often of irregular geometry with variable finite impedance. An appealing approach to handling this problem is to consider the local field around a particular ridge with the hope of splicing together a global solution from a set of local solutions. An initial study was made of the diffraction of a plane wave by a single cylindrical ridge of finite impedance. The method of matched asymptotic expansions (MAE), for large values of kR (where k is the wavenumber and R is the radius of curvature of the ridge), was employed to obtain a solution valid on, near, and behind the ridge. Predictions based upon this solution compared very well with the results of scale model experiments [J. Acoust. Soc. Am. Suppl. 1 **83**, S93 (1988)]. Recently, scale model experiments have been conducted to study the propagation of sound over two consecutive identical ridges. Preliminary results indicate that the MAE theory is indeed a powerful tool for the analysis of the diffraction of sound over complicated boundaries. [Work supported by NASA Langley Research Center.]

Session PPP. Speech Communication XIII: Recognition and Understanding (Poster Session)

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Contributed Papers

All posters should be set up before 2:00 p.m. All posters will be displayed from 2:00 to 5:00 p.m. To allow contributors the opportunity to see other posters, contributors of papers PPP1 to PPP12 will be at their posters from 2:00 to 3:00 p.m., contributors of papers PPP13 to PPP24 will be at their posters from 3:00 to 4:00 p.m., and contributors of papers PPP25 to PPP38 will be at their posters from 4:00 to 5:00 p.m.

PPP1. A categorical factor analysis of vowel distribution based on the modified qualification theory. Tetsunori Kobayashi and Toshiyuki Matsuda (Department of Electrical Engineering, Hosei University, 3-7-2 Kajino-cho, Koganei, 184 Japan)

Vowels in continuous speech have a very wide distribution in spectral space. However, if the phonemic environment is known, it is possible to compensate for the contextual variation in the spectrum and to make the distribution compact. In this paper, an attempt is made to determine what kind of categorical item (factor) should be considered to make the distribution compact using a modified qualification theory (MQT). Although original qualification theory has the fault in that it cannot deal with correlative relations between items, this MQT can do this because it has a self-organizing function for correlative relation. The results of an analysis of vowels in 492 words are as follows. To compensate for the variance, in /u/, it is most important to know whether the preceding sound is alveolar or not. In case the preceding sound is a labial, the place of the articulation of the next preceding sound is very important.

PPP2. Across-talker acoustic properties for place of articulation in nasal consonants. P. F. Seitz (INRS-Télécommunications, 3, place du Commerce, Ile-des-Soeurs, Verdun, Québec H3E 1H6, Canada), M. M. McCormick, I. M. C. Watson, and R. A. W. Bladon (Oxford University Phonetics Laboratory, 37/41 Wellington Square, Oxford OX1 2JF, England)

Several recent investigations of nasal consonants have focused upon the nature of the spectral change that takes place over the acoustic boundaries between nasal murmurs and vowels. Existing findings that use an algorithmic classification of place of articulation in nasals are based upon data from only a small numbers of talkers. A new method for parametrizing spectral change in nasal-to-vowel transitions was developed and was used to develop classification criteria that were applied to a larger talker base of 20 female and 20 male talkers who produced a total of 854 nasal-initial tokens. Male–female differences of a predicted kind (approximately 1 Bark spectral shift) were detected. However, the automatic classification criteria were sufficiently robust that, even ignoring such spectral shifts, the technique yielded approximately 79% correct classification of place of articulation in initial nasals. [Research supported by Alvey Project MMI 092.]

PPP3. Speaker-independent recognition of unvoiced plosives using a convex time pattern of *a posteriori* probability. Jouji Miwa (Faculty of Engineering, Iwate University, Morioka, 020 Japan)

This paper describes a method of speaker-independent recognition of unvoiced plosives. In this method, the convex characteristics of the time pattern of *a posteriori* probability is adopted for eliminating effects of speaker difference and coarticulation. The time pattern of *a posteriori* probability is more suitable than that of the distance pattern. In the first stage of the method, *a posteriori* probabilities for four categories (/p/, /t/, /k/, and silence) are calculated frame by frame from a five-channel spectrum of five time frames using the Bayes theorem. In the next stage, the convex part of the time pattern of *a posteriori* probability is decided as an unvoiced plosive. The decision using the dynamic characteristics is more suitable for speaker-independent recognition than that using static threshold. The recognition experiments are conducted for about 1400 samples of unvoiced plosives in 166 Japanese city words uttered by five male speakers. These experiments are carried out under the condition of automatic phoneme detection and without the knowledge of the following vowel. The recognition rate of 81% is obtained for the speaker-dependent case and 59% for the speaker-independent case. [Work supported by Grant-in-Aid for Scientific Research on Priority Areas, The Ministry of Education, Science and Culture of Japan.]

PPP4. Phonetic feature extraction based on mutual information. Noriyuki Aoki, Naoki Hosaka, and Katsuhiko Shirai (Department of Electrical Engineering, Waseda University, 3-4-1 Ohkubo, Shinjuku-ku, Tokyo, 160 Japan)

A novel method of feature extraction for phoneme recognition in continuous speech is proposed that employs mutual information between acoustic features and phonemes. Various acoustic features are coded by the vector quantization (VQ) method and a method to discriminate phonemes by the effective combination of these VQ codes is developed. To construct an optimal algorithm for phoneme discrimination, entropy and mutual information, in addition to conditional probability, between phoneme labels and features are also taken into consideration. The effectiveness of each acoustic feature for describing the characteristics of the phoneme in a given environment is evaluated based on the mutual information. The LPC mel-cepstrum, its pattern of temporal changes over frames, and power are used as acoustic features. Three experiments were

conducted. The first was on the optimization of the frame labeling. The second was on the detection of the vowel using a sequence of frame labels. The third was on word discrimination. The effectiveness of the proposed method was verified by these experiments.

PPP5. A method for selecting an optimum phoneme sequence using *a posteriori* probabilities of phonemes. Shozo Makino, Satoshi Moriai, and Ken'iti Kido (Research Center for Applied Information Sciences, Tohoku University, 2-1-1 Katahira, Sendai, 980 Japan)

A phoneme recognition system using phoneme discriminant filters was proposed, where segmentation and phoneme recognition were carried out simultaneously. However, the method had many phoneme insertion errors and often produced phoneme sequences that were inconsistent with Japanese phoneme connection rules. This paper proposes methods for reducing the number of phoneme insertion errors using inhibitory functions and for selecting an optimum phoneme sequence using dynamic programming based on the *a posteriori* probabilities of the phonemes. The definition of inhibitory functions is based on the Japanese phoneme connection rules and the phoneme duration information. This method was tested for the recognition of phonemes in a 212-work vocabulary uttered by ten male speakers. The inhibitory functions reduced the total number of phoneme insertion errors to 1/5 and, especially, reduced the number of consonant insertion errors. The *a posteriori* probabilities of phonemes were computed based on distributions of the output of the phoneme discriminant filters.

PPP6. Word recognition using a two-dimensional mel-cepstrum in noisy environments. Tadashi Kitamura and Etsuro Hayahara (Department of Engineering, Nagoya Institute of Technology, Gokiso, Showa-ku, Nogoya, 466 Japan)

This paper presents a word recognition method using a two-dimensional mel-cepstrum (TDMC) in noisy environments. The TDMC is defined as the two-dimensional Fourier transform of mel-frequency scaled log spectra in the frequency and time domains, and it consists of the average features and dynamic features of two-dimensional mel-log spectra. In comparison with the standard method using a one-dimensional mel-cepstrum, this method gives better recognition rates. Furthermore, the selection of the size of the TDMC used to describe dynamic features, the optimum weighting between the average and dynamic features, and reference noise patterns are taken into consideration in order to improve this method. Speaker-dependent word recognition experiments using 10 Japanese digits uttered by six male speakers indicate that recognition error rates lower than 2% can be obtained for noise-free speech and white-noise-added speech of 20 and 10 dB SNR.

PPP7. Automatic speech recognition of small vocabularies within the context of unconstrained input. R. W. Bossemeyer (AT&T Bell Laboratories, Naperville, IL 60566-7033), J. G. Wilpon, C. H. Lee, and L. R. Rabiner (AT&T Bell Laboratories, Speech Research Department, Office 2C-573, 600 Mountain Avenue, Murray Hill, NJ 07974-2070)

Speaker-independent recognition of small vocabularies, spoken over the long-distance telephone network has been demonstrated to be a viable technology. However, the algorithms tested and the tasks evaluated both assumed that user input would be restricted to only a small set of defined vocabulary words. Recently, a large scale trial of speaker-independent, isolated-word, speech-recognition technology was carried out in Hayward, California. The task chosen required that users speak, in isolation, one of five defined vocabulary words (*collect, calling-card, person, third-number*, and *operator*). Observations of customer responses during this trial, indicated that about 20% of the utterances had the desired vocabulary item along with extraneous input that ranged from nonspeech sounds to groups of words (e.g., "I want to make a *collect* call please"). Our current recognition algorithms have not been designed to handle this type of input. As such, a modification of the recognition algorithms had to be made to handle words embedded in speech (i.e., a form of key-word spotting). In this paper, two recognition algorithms are presented, one based on templates and the other based on hidden Markov models. Both algorithms are designed to recognize vocabulary words in the context of unconstrained speech. Currently, recognition rates of 99% for strictly isolated input (i.e., with no extraneous speech or gross artifacts) and 90% for vocabulary words spoken in unconstrained speech are being achieved.

PPP8. Distributed processing of finite state grammars for connected word speech recognition. Stephen C. Glinski (AT&T Bell Laboratories, 600 Mountain Avenue, Room 7C-427, Murray Hill, NJ 07974)

An algorithm is presented for performing real-time connected word speech recognition under the constraint of a finite-state grammar, and executed in a multiprocessor pattern recognition computer. Each processing element of the computer consists of a 40-MIP one-chip pattern recognition processor and its associated local memory. [S. Glinski *et al.*, "The Graph Search Machine (GSM): A VLSI Architecture for Connected Speech Recognition and Other Applications," Proc. IEEE **75** (9), 1172–1184 (1987)] Aspects of the algorithm discussed include: the choice of models for speech sound units, words, and grammars; the problems of partitioning the models across processing elements; and the problem of removing redundancy in the finite state grammar. Extensions of the algorithm to more powerful grammars is also discussed. The capacity of the recognizer (i.e., vocabulary size) as a function of grammatical complexity is presented. Results indicate that high-accuracy medium vocabulary speaker-trained connected word recognition is readily attainable.

PPP9. Acoustic segmentation and phonetic classification in the SUMMIT system. James Glass, Robert Kassel, David Kaufman, Michael Phillips, Stephanie Seneff, and Victor Zue (Room 36-575, Massachusetts Institute of Technology, Cambridge, MA 02139)

This paper describes recent work on acoustic segmentation and phonetic classification as part of an effort in speech understanding system development. The signal representation is based on an auditory model that incorporates known properties of the human auditory system, including critical-band filtering, saturation, adaptation, forward masking, and synchrony detection. Acoustic landmarks are determined using a measure of local similarity. These landmarks are embedded in a multilevel structure in which information ranging from coarse to fine is represented in an organized fashion. An analysis of the acoustic structure, using 500 utterances from 100 different talkers, shows that it captures over 96% of the acoustic-phonetic events of interest with an insertion rate of less than 5%. Phonetic classification is achieved by defining a set of generic property detectors based on the knowledge of acoustic phonetics. The settings of the parameters are obtained by a search procedure, using a large body of training data. The resulting features are selected through an optimization process. Classification is achieved using conventional pattern classification algorithms. Classification performance was evaluated against hand-aligned phonetic transcriptions provided by phoneticians. The evaluation was based on 225 sentences from 45 talkers using a set of 38 context-independent phone labels. The correct label was the top choice over 70%

of the time, and within the top three over 90% of the time. [Work supported by DARPA-ISTO under contact N00039-85-C-0254.]

PPP10. Sentence understanding of spoken Japanese using phrase spotting and dependency grammar. Tatsuji Ito and Sei-ichi Nakagawa (Faculty of Engineering, Toyohashi University of Technology, Tenpaku-cho, Toyohashi, 440 Japan)

In order to recognize spoken Japanese sentences, a speech understanding system that consists of four stages was constructed. First, the system hypothesizes syllable candidates using an HMM-based syllable spotter. Second, it concatenates the syllables to obtain word candidates, checking the tree-structured dictionary. Third, it connects word candidates to get phrase candidates using a Japanese phrase spotter. At this stage, a finite state automaton that represents the Japanese intra-phrase grammar is used. Finally, the system selects the best phrase sequence as a sentence recognition result using a backward parsing algorithm. This algorithm makes it possible to analyze the syntactic structure of a phrase sequence or a partial sentence using Japanese dependency (*kakari-uke*) grammar. Japanese sentences for the recognition task were inquiries on the UNIX system and spoken by six male speakers at natural speed. Syllables extracted from the continuous speech were used as training data for the HMM syllable spotter. The system's performance was evaluated from these speech data. The semantic analysis of this system is also described.

PPP11. A speech understanding system based on a topic-oriented language model. Masahiro Hori, Katsuhiko Tsujino, Riichiro Mizoguchi, and Osamu Kakusho (ISIR, Osaka University, 8-1 Mihogaoka, Ibaraki, 567 Japan)

Higher-level knowledge plays an important role in the speech understanding problem. A topic-oriented language model is used in SPURT-I (a SPeech Understanding System with Rule-based and Topic-directed architecture). SPURT-I consists of three subsystems: a rule-based acoustic analyzer SPREX (a SPeech Recognition EXpert), a word hypothesizer, and a topic-directed parser ASP (an ASsociated-based Parser). SPURT-I accepts Japanese speech divided into "bunsetus" (Japanese syntactic units), and attempts to output a corresponding word sequence. The basic assumption of this approach is that acoustically close phoneme sequences rarely correspond to semantically close words. Based on this assumption, SPREX recognizes a group of consonants such as /b,d,g,z/, /p,t,k/, and so on. As a result of this simplification, more candidate words are generated by the word hypothesizer than in the case of strict recognition. Then ASP reduces the effective number of candidates by associating the topic of utterance and the candidate words. Currently, SPURT-I has a 1000-word vocabulary for simple scenic descriptions. Experiments with actual utterances showed that nine sentences out of ten were recognized successfully and the "bunsetu" recognition rate was 97%. [Work partly supported by a Grant-in-Aid for Scientific Research, Ministry of Education.]

PPP12. Interactive problem solving with speech. Alexander I. Rudnicky, Robert A. Brennan, and Joseph H. Polifroni (Department of Computer Science, Carnegie-Mellon University, Pittsburgh, PA 15213)

Until recently, systems offering high-performance speaker-independent continuous speech recognition were not available, making it difficult to understand how speech should be used in interactive systems. The advent of the SPHINX system developed at Carnegie-Mellon University [K. -F. Lee and H. -W. Hon, Proc. IEEE ICASSP-88, 123–126 (1988)] has made it practical to address the issue of designing systems that integrate speech into "real-world" tasks. This paper describes experience with several tasks that use tightly coupled speech interaction: a programmable voice calculator, a personal scheduler, and a spreadsheet. The goal of this work is to create an environment that allows for the rapid prototyping of "spoken language" systems and allows the study of human factors issues that these entail. The environment that was developed includes the ability to rapidly configure and refine recognition systems using declarative specifications, and the ability to define control structures suitable to particular tasks. [Work supported by DARPA.]

PPP13. The Carnegie-Mellon Portable Speech Library. Fileno A. Alleva and Eric H. Thayer (Computer Science Department, Carnegie-Mellon University, Pittsburgh, PA 15213)

In order to promote the dissemination of research results and to help advance the general state of the art in speech recognition, a public domain library of plug-compatible subroutines, modules, and programs has been created. This paper describes the Carnegie-Mellon Portable Speech Library (CMPSL). CMPSL provides algorithms written in C that can be used to quickly prototype state-of-the-art speech recognition systems that are useful in their ability to perform the speech recognition task while also providing a basis from which to pursue further research. CMPSL's chief contributions are to make available recent advances in speech recognition at Carnegie-Mellon in the context of Lee's large-vocabulary, speaker-independent recognition system, SPHINX [K. -F. Lee and H. -W. Hon, Proc. IEEE ICASSP-88, 123–126 (1988)], and to suggest a framework within which speech researchers can make the results of their work available to other members of the automatic speech recognition community. The scope of CMPSL includes digital signal processing, training, recognition, and evaluation algorithms. [Work supported by DARPA.]

PPP14. Analysis, perception, and recognition of isolated Korean vowels. Hyun-yeol Chung, Shozo Makino, and Ken'iti Kido (Research Center for Applied Information Sciences, Tohoku University, 2-1-1 Katahira, Sendai, 980 Japan)

Distribution of the formant frequencies of eight isolated Korean vowels were analyzed and compared to the results of perceptual experiments. Recognition experiments were carried out based on these formant distributions. In the distributions of vowels on the $F1$–$F2$ plane, overlappings were found between the vowels [u], [o], [ɔ], and [ɯ]. Here, [ɛ] and [e] overlapped heavily as if they were the same vowel. Using perceptual domains defined by the results of listening tests, the dialectal differences and iterspeaker differences were investigated. In a recognition experiment, a speaker-independent recognition with the Bayes decision method was conducted for 1760 isolated vowels spoken by 20 speakers. A recognition rate of 81.2% was obtained with the formant frequencies $F1$ and $F2$, while the rate with the LPC cepstrum coefficients was only 76.4%. Normalization for eliminating interspeaker differences showed good results for vowels uttered by speakers of the same dialect. The highest recognition rate of 99.8% was obtained using a supervised learning method.

PPP15. Consonant discrimination using the formant coarticulation model. Shigeru Chiba (Pattern Information Sciences Division, Electrotechnical Laboratory, 1-1-4 Umezono, Tsukuba, 305 Japan)

Formant trajectories in the vicinity of consonants in continuous speech are observed to discriminate the place of articulation of the consonants. The coarticulation model [S. Chiba, Proc. IEEE ICASSP-86, Paper 49-10 (1986)] with a target formant frequency for consonants is developed based on the observation of the formant trajetories in natural VCV utterances. The coarticulation model proposed here is as follows:

$$F_b - F_t = a(F_p - F_t) + b(F_e - F_t) + c(F_f - F_t) + E,$$

where F_b, F_p, F_e, and F_f are the second formant frequencies at the centers and the consonant boundaries of the initial and final vowel segments in a VCV sequence, F_t is the target second formant frequency of the consonant, and E is the estimation error. The parameter values of a, b, c, and F_t are estimated by the method of least squares using actual values. Three coarticulation models for labials, alveolars, and velars are estimated and these models are applied to the discrimination of place of articulation of consonants in continuous speech. Experimental results show that the models for labials and alveolars can estimate the formant trajectories accurately and that a high-discrimination rate of 94% is obtained by comparing prediction trajectories with actual values. In the case of velars, however, the discrimination rate is low.

PPP16. Discrimination of stop consonants using a data-driven analysis. Hitoshi Iwamida and Shinta Kimura (Pattern Information Processing Laboratory, Fujitsu Laboratories, 1015 Kamikodanaka, Nakahara-ku, Kawasaki, 211 Japan)

The characteristics of stop consonants are time-varying. In traditional running spectrum analysis, the frames are not always synchronized with the events of speech. In this presentation, a new data-driven analysis method is proposed in which the frames are synchronized with the events to extract the features of the stop consonants accurately. In this method, a feature vector is extracted from four or five frames that are equally spaced in a consonant segment. A voiced stop segment is defined as the segment between the release and the point where power exceeds a threshold. A voiceless stop segment is defined as the segment between the release and the voice onset. In these discrimination experiments, 94.0% of the voiced and 96.3% of the voiceless stops were correctly discriminated. The speech database used for these experiments was the Japanese monosyllables (/b,d,g,p,t,k/ + /a,i,u,e,o/) uttered by 20 speakers. It was confirmed that analysis synchronizing with consonant segments is effective for stop consonant discrimination.

PPP17. The difference between acoustic and auditory parameter signals as a cue for phonetic segmentation and categorization. Hans G. Tillmann and Bernd Pompino-Marschall (Institut für Phonetik und Sprachliche Kommunikation der Universität München, Schellingstrasse 3, D-8000 Munich 40, Federal Republic of Germany)

Normally, speech recognition systems are based on purely acoustic or on auditorily modeled acoustic speech signals, respectively. The difficulties in segmentation and categorization encountered by those models may partly be overcome by a technique that exploits the difference signal between acoustic and auditory parameters. The difference signal tested here was between the sound-pressure envelope and the loundness time series. Sound-pressure level (dB-scaled) was calculated as the mean value of absolute amplitude within a rectangular widow of 300 sample points (= 15 ms). Loudness was computed every 15 ms according to Paulus and Zwicker [Acustica 27, 253–266 (1972)] by using critical band amplitudes approximated by averaged DFT values (with frequency-dependent differences in length of the Hanning window; < 500 Hz: 60 ms; 500–1500 Hz: 30 ms; > 1500 Hz: 15 ms). For calculation of the difference, signal loudness and sound-pressure level were normalized to $0 < x < 1$ and subtracted from one another.

PPP18. Automatic speech recognition based on findings of the human processes of speech perception. Hirofumi Udagawa, Sumio Ohno, and Hiroya Fujisaki (Department of Electronic Engineering, Faculty of Engineering, University of Tokyo, Bunkyo-ku, Tokyo, 113 Japan)

One of the shortcomings of coventional methods of ASR is that they operate on principles that are quite different and far less flexible than those of human speech perception. Previous studies on the perception of continuous speech have revealed that (1) a linguistic message may have multiple perceptual representations in various units which differ in size, and that (2) a representation of the input speech in larger units is derived earlier in the process of speech perception than that in smaller units such as phonemes. Hence, a word is often recognized on the basis of global characteristics without being subjected to detailed phonemic analysis. Based on these findings, a novel system for continuous speech recognition is proposed. The front end analyzes the input speech with various degrees of temporal and frequency resolution, and the results are matched against templates of various linguistic units that differ in size. The recognition of smaller units takes place only when the matching of larger units fails to produce a unique output. The capability of the approach was demonstrated by preliminary recognition experiments of weather forecast sentences in Japanese.

PPP19. Word recognition using synthesized reference templates. Mats Blomberg, Rolf Carlson, Kjell Elenius, Björn Granström, and Sheri Hunnicutt (Departments of Speech Communication and Music Acoustics, KTH, Box 700 14, S-100 44 Stockholm, Sweden)

A major problem in large-vocabulary speech recognition is the collection of reference data and speaker normalization. In this paper, the use of synthetic speech is proposed as a means of handling this problem. An experimental scheme for such a speech recognition system will be described. A rule-based speech synthesis procedure is used for generating the reference data. Ten male subjects participated in an experiment using a 26-word test vocabulary recorded in a normal office room. The subjects were asked to read the words from a list with little instruction except to pronounce each word separately. The synthesis was used to build the references. No adjustments were done to the synthesis in this first stage. All the human speakers served better as reference than the synthesis. Differences between natural and synthetic speech have been analyzed in detail at the segmental level. Methods for updating the synthetic speech parameters from natural speech templates will be described. [This work has been supported by the Swedish Board of Technical Development.]

PPP20. A telephone number information system for private branch exchanges. Shoji Hiraoka, Toshiyuki Morii, Masakatsu Hoshimi, and Katsuyuki Niyada (Matsushita Research Institute Tokyo Inc., 3-10-1 Higashimita, Tama-ku, Kawasaki, 214 Japan)

A telephone speech-recognition information system has been developed using a personal computer, a speech-recognizer, and a voice synthesizer. This speech recognizer is speaker independent and has a large vocabulary [S. Hiraoka *et al.*, IEEE ICASSP'86, pp. 2.8.1–4]. The user calls the system and gives the surname and division name of the addressee; both words are recognized and the extension number is supplied. Previously developed microphone phoneme templates were exchanged and made telephone compatible. The speech-sample data, which were used for making the previous templates were reused and converted to sound via a mouth simulator. The data were then returned via a handset to the computer through a PBX line. The new phoneme templates were validated using 18 individuals' calls of 212 words each, giving a recognition rate of 91.2%. The new effective templates were easily made because the data had been previously labeled. The recognizer produces 15 candidates for

each of the two words and the personal computer combines them in order of priority to improve the effective recognition performance.

PPP21. SAIPH: A segmentation system for automatic labeling of a large speech database. Application to speech recognition. Jean-Claude Junqua (Speech Technology Laboratory, 3888 State Street, Santa Barbara, CA 93105 and C.R.I.N.-C.N.R.S., Vandoeuvre les Nancy, France) and Hisashi Wakita (Speech Technology Laboratory, 3888 State Street, Santa Barbara, CA 93105)

Several attempts for automatic segmentation and labeling of large speech databases have been made. Those previous studies proposed to (1) align the input utterance with the manually labeled reference utterance using dynamic time warping, (2) segment and label the utterance into broad phonetic classes prior to time alignment, and (3) use trained reference patterns. In some cases, phonetic or heuristic knowledge has been used to refine the boundaries. This approach does not use reference units and thus can be applied also to speech recognition. A transition measure derived from the perceptually based linear predictive analysis (PLP) was defined. Basic segmentation is obtained by use of heuristic knowledge based on this transition measure in addition to the energy and the zero-crossings parameters. An evaluation on a keyboard database (104 words) spoken by two speakers (one male and one female) showed that more than 92% of the good segments are obtained after this stage. A broad classification, knowledge of the expected phonetic string, and phonological rules are applied to deal with the oversegmentation and undersegmentation for further improvement.

PPP22. Measures of phonological rule set coverage. Michael Cohen, Jared Bernstein, and Hy Murveit (Speech Research Program, SRI International, Menlo Park, CA 94025)

Some speech recognition systems use alternative lexical forms to cover various pronunciations encountered when people speak. However, excess forms in the lexicon degrade recognition performance. Thus lexicons should maximize coverage of actual pronunciations that occur, while minimizing the number of alternative pronunciations. A set of phonological rules for English has been developed at SRI to cover a large body of transcriptions of sentences read by American speakers. This phonological rule set covers 98.5% of the segments in a separate set of transcriptions. *Rule efficiency* can be defined as the ratio of the number of times forms generated by a rule were spoken in a corpus of test data to the number of forms generated by the rule for the sentences in that test data. By this definition, palatalization of /t/ and /d/ before tautosyllabic /r/ is efficient, while flapping of /t/ that follows /l/ is not. There is a trade-off between the coverage of a rule set and the number of alternative pronunciations generated by the rule set. Rule efficiency can be used to rank rules in a rule set, to derive a family of rule sets that maximize coverage for a given number of alternative forms represented, or to select among alternative rule sets. [Work supported by DARPA.]

PPP23. Relationships between phoneme recognition accuracy and sentence recognition accuracy. Yoshihisa Ohguro and Seiichi Nakagawa (Faculty of Engineering, Toyohashi University of Technology, Tenpaku-cho, Toyohashi, 440 Japan)

The relationships between phoneme recognition accuracy and sentence recognition accuracy for continuous speech recognition using a simulated phoneme recognizer are described. A left-to-right and top-down parser based on Earley's algorithm was used as a syntactic parser. The beam search technique was embedded in this parser to reduce the search space. The syntactic constraints were represented by a context-free grammar. The word lattice for an utterance was generated by a word spotting algorithm from an ambiguous phoneme sequence. The input to the parser was the word lattice of spotted words that consisted of a group of four terms: word name, beginning point, ending point, and matched score. Three tasks, "connected word (digits or city names)," "computer network (about 250 words)," and "inquiries on UNIX (about 500 words)" were used for the evaluation. From experiments, the conclusions are as follows. (1) It is difficult to recognize continuous speech without syntactic knowledge, (2) The search space ought to be large for managing an ambiguous input. (3) The sentence recognition rate is not so sensitive to vocabulary size.

PPP24. A frame-based system for deep understanding: Toward understanding puns and metaphors. Masuzo Yanagida and Osamu Takizawa (Communications Research Laboratory, 4-2-1 Nukui-kitamachi, Koganei, 184 Japan)

An approach to deep understanding of speech is proposed. Deep understanding here means detection and analysis of hidden meanings like puns and metaphors. This report focuses mainly on understanding puns that involve words composed of the same number of syllables of different or similar phonemes. The analysis procedure for puns is evoked if some of the phoneme candidates of a high confidence factor in the input phoneme lattice differ from those in the phoneme sequence of legitimate words in the dictionary even if the input phoneme sequence seems understandable using low rank phonemes. The system searches possible words in the dictionary and assigns them scores according to the degree of coincidence between the phoneme sequence in the dictionary and that in the input phoneme lattice evaluating semantic conformity. In case there exist some words with high fitting scores, the system concludes that the input sentence has a hidden meaning assuming the doubly interpreted phoneme sequence to be a pivot word. [Work partly supported by Science and Technology Agency, Japan.]

PPP25. Benchmark test procedures for automatic speech recognition systems. David S. Pallett (Room A, 216 Technology Building, National Bureau of Standards, Gaithersburg, MD 20899)

Benchmark test procedures have been developed and implemented in the DARPA-sponsored program in automatic recognition of continuous speech. These tests make use of a database of recorded speech that includes material for both speaker-dependent and speaker-independent technologies. In implementing these uniform tests, it is necessary to define a standard lexicon of units to be scored and a standard or reference orthographic representation for each sentence, and to select a scoring procedure and error taxonomy. Considerations taken in identifying the reference lexicon and representations will be outlined. For the DARPA program, a dynamic-programming string alignment procedure was implemented, and errors were defined in terms of substitutions, insertions, and deletions. This scoring software is publicly available. Known limitations of this procedure will be discussed. [Work supported by DARPA.]

PPP26. Acoustic properties for the distinction of Japanese stop consonants. Keikichi Hirose (Department of Electronic Engineering, University of Tokyo, Bunkyo-ku, Tokyo, 113 Japan)

Of all the sounds of speech, stop consonants present the greatest obstacles to automatic speech recognition. A study has been conducted to find acoustic properties that can be exploited for reliable distinction among the stop consonants of Japanese in CV utterances produced by five male speakers of Japanese. Short-time spectra were measured at the stop burst and at the onset of the following vowel. It was found that the most consistent property for distinguishing velar stops from other consonants is the presence of a dominant peak in the burst spectrum in the vicinity of the second/third formant of the following back/front vowel. On the other hand, the distinction between alveolar and labial stops can be made on the basis of the second formant transition when the stops are followed by a back vowel. This transition is concave for alveolars and almost level or convex for labials. When these stops are followed by a front vowel, however, the overall slope of the burst spectrum can be more reliably used for their distinction than the second formant transition. A similar analysis was also made on the nasal consonants of Japanese. [Work conducted at Research Laboratory of Electronics, Massachusetts Institute of Technology while the author was on leave from University of Tokyo.]

PPP27. Speech recognition using least-squares models for the contextual variations in the acoustic characteristics of phonetic segments. Yoshiharu Abe and Kunio Nakajima (Information Systems and Electronics Development Laboratory, Mitsubishi Electric Corporation, 5-1-1 Ofuna, Kamakura, 247 Japan)

This paper describes a speech recognition method using linear models to predict the variations in a feature vector caused by the phonemic context. The feature vector in a phonetic segment is decomposed and formulated by the sum of a context-independent vector, a context-dependent vector, and an estimation error. The second component is given by the product of a weight matrix and an environment vector. Least-squares estimations of the context independent vectors and the weight matrices for top-down, bottom-up, and mixed models are obtained algebraically. The top-down model obtains the environment vector from the phoneme string transcribed in the lexicon, while the bottom-up model obtains it from the input feature vectors. Speaker-dependent recognition experiments were carried out using an open vocabulary of 200 words spoken by two male and two female speakers. The average word error rate of 19.1% without any contextual models was reduced to 7.4%, 5.3%, and 3.3% by the top-down, bottom-up, and mixed models, respectively.

PPP28. On the use of triphone models for continuous speech recognition. Kai-Fu Lee (Department of Computer Science, Carnegie-Mellon University, Pittsburgh, PA 15213)

This paper describes both completed and future work with triphone models for speech recognition. Triphone models are a powerful subword modeling technique because they account for the left and right phonetic contexts. One problem with triphone models is that there are too many models for the amount of training data available. Moreover, many triphone contexts are very similar. In view of this, *generalized triphones* are introduced, which are created by clustering similar triphones together using an information theoretic criterion. This technique provides a way of finding the "right" number of models for any task. With generalized triphone models, a 96% accuracy with a 1000-word speaker-independent continuous task was achieved. There are only 2500 triphones for the above task. For a different task, there will be new triphones; for a natural English task, the number of triphones will be much larger. In the future, improvement of the applicability of triphone modeling could be done by (1) training a large set of generalized triphone models for English, and (2) exploring techniques to deal with new triphones. [Work supported by DARPA.]

PPP29. Automatic discovery of acoustic measurements for phonetic classification. Michael S. Phillips (Room 36-575, Massachusetts Institute of Technology, Cambridge, MA 02139)

One approach to phonetic classification is to rely on the use of well-motivated acoustic measurements such as formant frequencies. However, the extraction and the use of these measurements are often heuristically based, thus resulting in fragile classification performance. Alternatively, one can rely completely on learning procedures to automatically discover the phonetic regularities. But the requirement on computation and training data may become prohibitive. This paper describes an approach in which the determination of the acoustic measurements is formulated as a constrained search problem. In the present investigation, the constraints are provided through a set of generic measurement primitives based on knowledge of acoustic phonetics. These primitives, such as spectral moments or energy changes, have free parameters that control the specifics of the measurements. Primitives may be combined to form more complex measurements according to a set of constraints. The set of primitives, along with their free parameters and the combination constraints, specify a space of allowable measurements. This space can be searched using a measure of phonetic discrimination performance on a large body of training data to obtain an optimal set of measurements for a particular task. Results of classification experiments using these automatically generated measurements will be presented for various phonetic discrimination tasks. [Work supported by DARPA-ISTO under contract N00039-85-C-0254.]

PPP30. Recognition of speech produced in a simulated noisy environment. Hiroaki Hattori and Kazunaga Yoshida (C & C Information Technology Research Laboratories, NEC Corporation, 4-1-1 Miyazaki, Miyamae-ku, Kawasaki, 213 Japan)

Pronunciation manner varies depending on the noise of the speech environment. This is known as the Lombard effect. It affects the acoustic features of speech such as intensity, pitch, duration, and spectral shape. Because vowels play an important role in Japanese speech recognition, special attention was paid to the variability of the acoustic features of vowels uttered in noise. First, the variability of the five Japanese vowels was examined using speech uttered while hearing noise through headphones. It was observed that the higher formants are unstable and that the energy in the middle frequency range of the vowel spectrum increases depending on the noise level. Based on these observations, a normalization method combining band limitation and spectral tilt compensation was proposed, and its effectiveness in the recognition of vowels was shown by experiment. Next, a new word recognition method utilizing this normalization technique was proposed. This method performs vowel normalization at the portion where both test and reference patterns are vowel-like sounds. The experimental results show the significant effectiveness of the proposed method. The recognition rate obtained by the proposed method with normalization is 96% for 50 words uttered by two male speakers, while the recognition rate without normalization is 80%.

PPP31. Recursive partitioning and parametric transformation for isolated digit recognition. Les Atlas, Carlos Rivera (Department of Electrical Engineering, FT-10, University of Washington, Seattle, WA 98195), and R. Douglas Martin (Department of Statistics, GN-22, University of Washington, Seattle, WA 98195)

Recursive partitioning [Breiman, Friedman, Olshen, and Stone, *Classification and Regression Trees* (Wadsworth, Belmont, CA, 1984)] is a trainable classification technique that has proven useful in a variety of applications. Some advantages of this technique include very fast recognition (due to tree architecture), the flexibility of a nonparametric estimator, and the ability to self-organize partitions that may illuminate the underlying structure of the data. Due to the special nature of the speech recognition problem, it was necessary to adapt the basic concepts of recursive partitioning. For example, in order to add invariance to differences in speaking rate, a parametric transformation [Atlas *et al.*, Proc. IEEE

Conf. Neural Inf. Proc. Sys. (1988)] was used to preprocess the speech signal. This parametric transformation converted a sequence of order N LPC parameters into a trajectory in N-dimensional space, where a sequence of samples of this trajectory was used to train the classifier. Observations of these trajectories confirmed the insensitivity to changes in the time axis and also showed that the greatest changes in the trajectory occurred during transient portions of the speech. These techniques are currently being integrated into a study of comparative performance for recognition of isolated digits. [Research supported by NSF and U. S. West Advanced Technologies.]

PPP32. Approach to speaker-independent connected word recognition for telephone application. Kei Miki (Research Laboratory, OKI Electric Industrial Company, Ltd., 550-5 Higashi-asakawa-cho, Hachioji, 193 Japan)

This paper describes a new speaker-independent connected word recognition method for telephone use. For this recognition, it is necessary to deal with the following four problems: utterance speed variations in speakers; spectrum variations caused by a speaker's sound source and vocal tract characteristics; interference between words in continuous speech; and telephone circuit noise. For the first problem, a new nonlinear time division method is proposed using the spectral transition ratio. The result shows a phoneticlike segmentation. This method is more effective than DTW (dynamic time warping) in the case of speaker-independent systems. For the rest of the problems, a set of features will be used that give a maximum recognition accuracy, for example, the spectral local peak, the ratio between the higher and lower frequency components, and spectral transition vectors. The method is quite robust in the face of spectral changes because the features cooperate with each other. This method shows a high performance in speaker-independent connected word recognition for telephone use.

PPP33. Word-boundary identification in automatic continuous speech recognition. Jonathan Harrington, Gordon Watson, and Maggie Cooper (Centre for Speech Technology Research, University of Edinburgh, Edinburgh EH1 1HN, Scotland)

Phoneme sequence constraints, such as /b l g/, that occur across word boundaries (*considerable growth*) but not word internally were used for word-boundary identification from continuous speech transcriptions. All three-phoneme, word-boundary sequences were derived by pairing P# (all word-final phonemes) with #PP (all word-initial, two-phoneme sequences) and PP# with #P in a 23 000-word lexicon including 70 000 reduced phonological forms. The resulting P#PP and PP#P sequences were matched against the same lexicon to determine which of them were excluded word internally. All boundary sequences excluded word internally were compiled into a tree structure that was matched against three sets of phonemic transcriptions corresponding to "fast," "normal," and "citation" speech production styles of 145 utterances. The results showed a word-boundary detection rate of between 39% and 50% in the three transcription sets. The incorporation of phoneme sequence constraints also reduces the number of word parsings compared with an earlier left-to-right chart-parsing approach for word-boundary identification [J. M. Harrington and A. Johnstone, Comput. Speech Lang. (1988), in press].

PPP34. A continuous speech recognizer using augmented continuous DP matching and its application. Ichiro Ogawa and Sei-ichi Nakagawa (Faculty of Engineering, Toyohashi University of Technology, Tenpaku-cho, Toyohashi, 440 Japan)

It is important to develop a real-time continuous speech recognizer for investigating the man–machine interface through voice. From this point of view, a continuous speech recognizer that employs general digital processors and an augmented continuous DP matching (DTW) algorithm was developed. This equipment consists of an FFT processor for feature parameter extraction, 4 ImPPs (Image Pipelined Processor, NECuPD7281) for DP matching, and a host computer (NEC PC-9801E) for handling the syntactic constraints. It can be applied to an arbitrary task that is composed of 140 reference patterns and is represented by a finite state automaton with 50 states. Additionally, the method of expression of the automaton for a given task and the semantic extraction to derive the semantic information from the recognition results were designed to improve the flexibility of the user interface. The recognizer with this user interface was applied to the Japanese-LOGO (vocabulary size: 104 words). In speaker-dependent recognition tests, the recognizer achieved a sentence accuracy of about 90%.

PPP35. Morphemic and syntactic analyses in a Japanese dictation system. Akinori Ito, Yoiti Ogawa, Michio Okada, Shozo Makino, and Ken'iti Kido (Research Center for Applied Information Sciences, Tohoku University, 2-1-1 Katahira, Sendai, 980 Japan)

A Japanese dictation system is being developed for task-independent continuous speech recognition. To achieve task-independency, syntactic and semantic constraints are used in the morphemic and syntactic analysis subsystems. The structure of the Japanese sentence is effectively described by a two-level grammar that consists of an intra-Bunsetsu part and an inter-Bunsetsu part. The Bunsetsu structure consists of one content word followed by several function words. Because most of the function words are short, they cannot be spotted individually with high accuracy. In the morphemic analysis subsystem, a *syntax-driven continuous DP matching algorithm* spots Bunsetsu-like units and generates a Bunsetsu lattice. In the spotting process, function words following a content word are predicted using the intra-Bunsetsu grammar. In the syntactic analysis subsystem, a Bunsetsu lattice is parsed by an algorithm based on the CYK algorithm. Each Bunsetsu candidate has syntactic markers. The inter-Bunsetsu grammar is expressed as the correspondence between a FEATURE marker in the modifier-Bunsetsu and a SLOT-FILLER marker in the head-Bunsetsu. A recognition rate of 81.5% was obtained from simulator-generated phoneme sequences with an 85% phoneme recognition score.

PPP36. Speech interface in a dialog system. Yutaka Kobayashi and Yasuhisa Niimi (Computer Science Laboratory, Kyoto Institute of Technology, Matsugasaki, Sakyo-ku, Kyoto, 606 Japan)

A speech interface for a database is being developed. The system tries to interpret sequences of natural language queries from a user by utilizing a wide range of knowledge sources. The acoustic analyzer transforms the speech signal into a phonetic lattice using the phonetic HMM-based analysis. The syntactic and semantic analyzer predicts those words that might appear next to partial sentence hypotheses. Instead of matching word by word, the unit of matching against the lattice is a part of concatenated word templates bounded by robust phones for dealing with coarticulation effect across word boundaries. The dialog component includes a user model and a topic management mechanism, so as to keep track of the user's intention and to alternate the conversation initiative between the system and the user. Knowledge sources are being refined and statistics are being gathered for a set of queries on sightseeing, where the vocabulary size is 500 × 1000 and the speech corpus contains 120 sentences from each of six speakers. Recent progress is reported in this paper.

PPP37. An isolated spoken word database using CD-ROMs. Shozo Makino, Masato Abe, and Ken'iti Kido (Research Center for Applied Information Sciences, Tohoku University, 2-1-1 Katahira, Sendai, 980 Japan)

A database of isolated spoken words has a 212-word vocabulary uttered by 40 male and 30 female speakers and further a 3000-word vocabulary uttered by 10 male and 20 female speakers. The 212-word vocabulary was selected to contain most of the Japanese phoneme contexts. The 3000-word vocabulary is a set of Japanese railroad station names. The speech data were sampled at 24 kHz and digitized to 12 bits. About 1000 magnetic tapes were used for the recording of the speech data. However, because of the huge number of the magnetic tapes, it is very difficult to deliver these tapes to speech researchers, and there are many problems in maintenance. For building up a database, the CD-ROM has many merits such as compactness and price. Furthermore, the so-called High Sierra Format was published last year and most computers can access CD-ROMs. The magnetic tapes are being converted to CD-ROMs in that format. By April 1989, two CD-ROMs will be available that will contain the 212-word vocabulary uttered by 50 speakers.

PPP38. Word boundary placement in faintly perceived speech. Anne Cutler and Sally Butterfield (MRC Applied Psychology Unit, 15 Chaucer Road, Cambridge CB2 2EF, United Kingdom)

The recognition of words in continuous speech is made more difficult by the absence of explicit word boundary markers. Cutler and Norris [J. Acoust. Soc. Am. Suppl. 1 **77**, S39 (1985)] proposed that listeners compensate for the absence of explicit word boundary cues by adopting a strategy of assuming each strong syllable to be a potential word onset. Cutler and Carter [J. Acoust. Soc. Am. Suppl. 1 **81**, S67 (1987)] showed that the structure of the English vocabulary is conducive to the application of such a strategy. In the present experiment, listeners' speech reception thresholds were individually estimated; the listeners were then presented with short unpredictable utterances (e.g., "never just convict them") close to this threshold. Exactly as predicted by the proposed segmentation strategy, erroneous boundary insertions occurred more often before strong syllables (i.e., syllables containing a full vowel) than before weak syllables, while erroneous boundary deletions occurred more often before weak syllables (i.e., syllables containing a reduced vowel) than before strong syllables. Word frequency analyses showed that this result was not an artifact of a bias towards high-frequency responses. [Research supported by the Alvey Directorate, U.K.]

FRIDAY AFTERNOON, 18 NOVEMBER 1988 — MAUI ROOM, 2:00 TO 5:00 P.M.

Session QQQ. Underwater Acoustics X: Scattering in Underwater Acoustics

Ralph Stephen, Cochairman
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Contributed Papers

2:00

QQQ1. Resonance and directionality of linear arrays of monopole scatters. I. Tolstoy (Knockvennie, Castle Douglas DG7 3PA, Scotland) and Alexandra Tolstoy (Naval Research Laboratory, Washington, DC 20375)

Linear arrays of n identical resonators (bubbles, gas-filled balloons, or very thin shells) spaced a distance l apart and insonified by a plane wave at frequencies close to or below the monopole resonance (the radial "bubble" frequency), display resonances due to multiple scatter interaction. Whereas such systems exhibit true normal modes of vibration (*superresonances*) only near elastic boundaries in a fluid, they exhibit strong partial resonances (*quasiresonances*, or QRs) in fullspaces, for both finite and infinite n, with absolute amplifications relative to the free-field amplitude of up to 7×10^2 inside and upon the surface of the resonator. These QR effects have *strong directionality*: an important result of the multiple scatter interaction is to make the response of individual elements—their effective monopole strengths—very sensitive to the direction of arrival of the plane wave. Theoretical directionality factors of the order of 200 are achieved in this way prior to any form of phasing, filtering, or processing of the monpole outputs *qua* elements of an array.

2:12

QQQ2. Scattering and reflection of acoustic waves by striated surfaces depicting the Arctic Ocean ice cover. Theodore P. McClanahan and O. I. Diachok (Radix Systems, Inc., 2 Taft Court, Suite 203, Rockville, MD 20850)

Acoustic reflectivity from plastic models of sea ice ridges on a thin plate were investigated for $0.67 < ka < 101$ (where k is the wavenumber and a is the mean ridge depth). Models were constructed of lucite, which has acoustic velocities consistent with analytically estimated velocities of sea ice ridges and of "fast cast," a material with much lower velocities. For $ka > 1$ measurements from randomly oriented and parallel lucite and fast cast ridges on a thin plate and on an infinite half-space were nearly identical, and in good agreement with Twersky's theoretical predictions. At $ka < 1$ and at grazing angles less than 20°, measured reflection losses were significantly greater than predictions: twice as large for randomly oriented lucite ridges on a thin plate. The phase of the reflected signal was strongly dependent on the orientation and the material property of the ridges. Results for randomly oriented lucite ridges were found to be consistent with previously reported inferences of reflection loss and phase based on matched field processing of actual Arctic data and consistent with reported resonance effects.

2:24

QQQ3. Wave scattering from rough surfaces using the facet-ensemble approach. Yushieh Ma, Ruyen Ro, and Vasundara V. Varadan (Department of Engineering Science and Mechanics, The Pennsylvania State University, University Park, PA 16802)

The facet-ensemble method has been widely used to investigate scattering from long crested surfaces whose roughness and slopes are, in general, fairly small. In addition to its simple physics, this method also takes the full advantage of modern computing facilities to provide numerical results in a very efficient manner under any change of surface conditions. In order to extend this method to any irregular surfaces of different roughness and slope scales, multiple scattering as well as wedge diffraction should be carefully incorporated. Therefore, a modified facet-ensemble approach that does not restrict the grazing angles is reported here. In order to describe the scatttering characteristics of random rough surfaces, the statistical functions can be built, but have not been considered before, into the formalism that saves even more CPU time in performing the ensemble averaging. A comparison is made using this approach with the experiment conducted on scattering from random Gaussian surfaces. Results agree very well and also presented is the effort towards the two-dimensional random rough surface scattering. [Work supported by the Research Center for the Engineering of Electronic and Acoustic Materials.]

2:36

QQQ4. Multiple scattering of radiation by rows of parallel cylindrical obstacles in water. Kazuhisa Matsuda, Toshiaki Kikuchi, Sumio Takahashi, Akio Hasegawa (The National Defense Academy, Yokosuka, 239 Japan), and Toshiyuki Nakanishi (Japan Marine Science and Technology Center, Yokosuka, 237 Japan)

Experimental forward scattering pressures for a line array of ten cylinders of various sizes and materials were obtained to compare with calculated values. An investigation was made for kd from 1 to 10 (k is the wavenumber and d is the center-to-center spacing) and $ka < 1$ (a is the cylinder radius). The calculations were made by Twersky's theory of multiple scattering. When kd is larger than 3, the pressure level of the second-order scattering wave is 10–20 dB lower than that of the first-order scattering wave. In this range, the experimental results agree well with the calculated results. Large discrepancies are found, however, for values of kd smaller than 2. Also discussed are the effects of boundary conditions.

2:48

QQQ5. Simple approximate formulas for backscattering of sound by spherical and elongated objects. T. K. Stanton[a] (Department of Geology and Geophysics, University of Wisconsin, Madison, WI 53706)

The scattering of sound by objects is a complex process and is dealt with both analytically and numerically. Regardless of method, calculation of the exact scattered field is quite laborious and may require much computer time as well as human time to program the computer. In order to simplify calculations of the backscattering by a fluid sphere, Johnson combined low- and high-frequency limits in a heuristic manner to obtain a simple closed-form solution (named the "high-pass" solution) [J. Acoust. Soc. Am. **61**, 375–377 (1977)]. In this present work, his approach is refined and extended to the prolate spheroid, straight finite cylinder, and bent finite cylinder. The high-pass solutions are formulated in a general way so as to describe scattering by idealized objects where their shape is simple (i.e., sphere, spheroid, etc.) and their material is lossless and by nonideal more realistic objects where their shape may be irregular and their material lossy. Comparisons are made of the scattering of these objects using both modal series solutions and the "high-pass" solutions for fluid, elastic, rigid and fixed, and gaseous materials over a wide range of frequencies. In addition to the numerical simulations, some of the high-pass solutions and corresponding modal series solutions are compared with data involving "ideal" objects (machined Dural) and "nonideal" objects (marine organisms: shrimp, euphausiids, and fish). The numerical and experimental results show promise for use of the high-pass models for quick estimates of backscattered sound of many types of objects. [Work supported by the ONR.] [a] Present address: Woods Hole Oceanographic Institution, Woods Hole, MA 02543.

3:00

QQQ6. Benchmark comparisons of underwater acoustic scattering models. Guy V. Norton, Michael F. Werby, and Michael D. Collins (Naval Ocean Research and Development Activity, Stennis Space Center, MS 39529)

Benchmark calculations for waveguide scattering problems in realistic ocean environments will be presented. The three models used in the study are based on Huygens principle [M. F. Werby and R. B. Evans, IEEE J. Ocean. Eng. **OE-12**, 380–394 (1987)], a projection approach [G. V. Norton and M. F. Werby, Math. Comput. Model. **11**, 81–86 (1988)], and the method of matched asymptotics [M. D. Collins and M. F. Werby, J. Acoust. Soc. Am. (submitted)]. The first two methods utilizes the normal mode solution for a point source in a waveguide along with the T matrix that describes scattering from an axially symmetric object in free space, in order to obtain the scattered field near the object. The method based on Huygens principle uses this scattered nearfield, along with the Green's function that satisfies the waveguide's boundaries to solve for the scattered field far from the object. The projection method, however, projects this scattered nearfield onto the directional multipole solution for the waveguide and the resulting field can then propagate out into the waveguide. The asymptotic method involves the parabolic equation method and is based on a simple approximation of the waveguide Green's function in the nearfield. This method does not require a decomposition of the incident field, which results in computational efficiency. [Work supported by the Naval Ocean Research and Development Activity.]

3:12

QQQ7. Exact solutions for sound diffraction by truncated wedges. I. Tolstoy (Knockvennie, Castle Douglas DG7 3PA, Scotland)

A previously described method for obtaining exact, explicit solutions for the diffraction by truncated wedges [I. Tolstoy, J. Acoust. Soc. Am. Suppl. 1 **82**, S61 (1987)] was valid for arbitrary vertex angles θ_w, θ'_w providing these were irrational fractions of π. This has now been extended to the case where these angles can be rational fractions of π. As before, the method combines exact, explicit representations of the field diffracted by a single line vertex with the classic multiple scatter self-consistent algorithm to formulate the interaction between two vertices. Thus there is a general exact and explicit solution for the problem of diffraction of a plane wave by a truncated wedge of any shape. This has been applied to the case of a thick rectangular sound barrier, in the wavenumber range $0.2 \lesssim kl \lesssim 2.8$, where l is the barrier thickness, to calculate the sound field in the shadow. The results are seen to agree well with published experimental results [Medwin *et al.*, J. Acoust. Soc. Am. **72**, 1005–1013 (1982)]. The method is quite general and may, in principle, be extended to more complicated polygonal and broken-line profiles with any number of vertices. [Work supported by ONR.]

3:24

QQQ8. A generalized wedge assemblage method for surface scattering calculations. R. S. Keiffer (NORDA, Numerical Modeling Division, NSTL, MS 39529) and J. C. Novarini (SYNTEK Engineering & Computer Systems, Rockville, MD 20852)

The facet ensemble or wedge assemblage (WA) method, which was originally developed for and successfully applied to long crested 3-D surfaces [J. C. Novarini and H. Medwin, J. Acoust. Soc. Am. **64**, 260–268 (1978); H. Medwin and J. C. Novarini, J. Acoust. Soc. Am. **69**, 108–111 (1981); Kinney *et al.*, J. Acoust. Soc. Am. **73**, 183–194 (1983)], has been

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extended to anisotropic 3-D surfaces. This allows for the modeling of scattering from more realistic surfaces and may give insight into the relative roles of diffractions and reflections from 3-D rough surfaces. In this new approach, the surfaces are decomposed into elemental finite wedges made up of triangular facets. In the model, each of these wedges has associated with it a diffracted response and, depending upon its orientation relative to the source and receiver, a reflection. The diffracted response of each finite wedge is calculated by appropriately scaling the response of an equivalent infinite wedge and windowing it by the time interval associated with the finite wedge. Reflections are handled separately and are determined by geometry and the relative sizes of the Frensel zone to the reflecting area. The impulse response for the entire surface is then the accumulation of the responses from the assemblage of finite wedges that make up the rough surface. As a preliminary application, the method is used to model scattering from 3-D wind-driven sea surfaces. Results are compared with those obtained from a Helmholtz–Kirchhoff based model.

3:36

QQQ9. Acoustic scattering from surfaces generated by nonstationary, nonuniform wind fields. J. C. Novarini (SYNTEK Engineering & Computer Systems, Rockville, MD 20852), R. S. Keiffer (NORDA, Numerical Modeling Division, NSTL, MS 39529), J. W. Caruthers (NAVOCEANO, Geophysics Department, NSTL, MS 39529), and G. Kerr (NORDA, Numerical Modeling Division, NSTL, MS 39529)

In an earlier paper [Novarini *et al.*, J. Acoust. Soc. Am. Suppl. 1 **83**, S48 (1988)], the scattered acoustic fields resulting from fully developed (generated from the Pierson–Moskowitz spectrum) and fetch-limited (generated from the JONSWAP spectrum) 3-D wind-driven sea surfaces were compared. Within the context of the Helmholtz–Kirchhoff theory, it was demonstrated that there are significant acoustic differences between these two descriptions of the sea surface. In this meeting, the scattered acoustic field resulting from 3-D sea surfaces generated from nonstationary, nonuniform wind fields are considered. These sea surfaces are generated from a spectral representation which, given the wind field specified over an extended area, is provided by the Global/Regional Deep Water Wave Model (D. Ressio, "Program Wave," Offshore & Costal Technologies Inc., Vicksburg, MS). This model has the ability to describe very realistic sea surfaces that include swell from distant storms and growing wind-driven seas. The scattered field resulting from such surfaces will be compared with results from sea surfaces generated by simple spectral descriptions which are functions of wind speed or wind speed and fetch. Preliminary results show strong differences in the coherence as well as the scattering strength over the entire frequency range under consideration (100–2000 Hz).

3:48

QQQ10. First- and second-order smoothing method for rough surface scattering. Akira Ishimaru (Department of Electrical Engineering, FT-10, University of Washington, Seattle, WA 98195)

There have been several recent investigations on rough surface scattering. They are directed at removing restrictions on the conventional perturbation and Kirchhoff approximations and also to devise a theoretical model to explain recent experimental data on backscattering enhancement. This paper presents a theory based on the diagram method of Bass *et al.* and the smoothing method of Watson and Keller. The rough surface is represented by an equivalent flat surface as was done by Bass–Fuks and Watson–Keller. The first-order smoothing solution is obtained from the Dyson and Bethe–Salpeter equations. The coherent field is identical to the Watson–Keller solutions. The first-order solution is close to Ito's solution, but it is reciprocal. The second-order smoothing solution has two terms. The ladder term is close to Ito's but the cross term exhibits some backscattering enhancement. Numerical examples for the one-dimensional Dirichlet surface are shown and compared with related theoretical calculations. [Work supported by NSF and ARO.]

4:00

QQQ11. Coupling finite elements and Helmholtz integral equations for solving radiation and scattering problems. Bernard F. Hamonic,[a] Oscar B. Wilson, and Steven R. Baker (Physics Department, Code 61, Naval Postgraduate School, Monterey, CA 93943)

Radiation and scattering problems encountered in fluid–structure interaction can be solved through a total finite-element formulation, or through a mixed formulation. In the first case, both the interacting body and the acoustic medium are modeled using finite elements. This approach leads to a linear equation system of large size when, for example, the overall dimensions of the transducer being analyzed are large compared with the wavelength of the radiated or scattered waves. In the second case, the body is modeled with finite elements and the fluid domain is implicitly represented through an integral equation. This completely eliminates the need for modeling the fluid, drastically reducing the size of the system to be solved. In this paper attention is focused on one combination for the mixed formulation, namely the coupling of the finite-element code ATILA [J. N. Decarpigny *et al.*, J. Acoust. Soc. Am. **78**, 1499–1507 (1985)] and the Combined Helmholtz Integral Equation Formulation code CHIEF [H. A. Schenck, J. Acoust. Soc. Am. **44**, 41 (1968)]. Results of test cases for both axisymmetrical and 3-D structures are presented. [Work partially supported by DRET.] [a] Permanent address: ISEN, 41, Boulevard Vauban, 59046 Lille Cedex, France.

4:12

QQQ12. Scattering amplitudes for rough interface scattering. David H. Berman (U.S. Naval Research Laboratory, Code 5160, Washington, DC 20375-5000)

A new exact expression for the scattering amplitudes in the case of fluid–solid interfaces is presented. This expression has the virtue of being manifestly reciprocal in the incoming and outgoing horizontal wavenumbers and is, therefore, a good starting point for making physically motivated approximations.The expression is a quadratic functional of various surface fields, but does not explicitly involve any Green's functions. In the limiting case of pressure release surfaces, the exact expression reduces to an expression derived by Dashen, and leads to a Kirchhoff-like approximation that agrees well with exact results over all scattering angles. Similar approximations for fluid–solid rough interfaces will be compared to exact calculations of the scattering amplitudes. [Work supported by NRI.]

4:24

QQQ13. Rough surface scattering with the parabolic wave equation using a split-step method. Eric I. Thorsos, John W. Ballard, and Terry E. Ewart (Applied Physics Laboratory, University of Washington, Seattle, WA 98105)

The split-step algorithm introduced by Tappert and Nghiem-Phu [J. Acoust. Soc. Am. Suppl. 1 **77**, S101 (1985)] for solving the parabolic wave equation (PE) with 1-D rough surfaces subject to the pressure release boundary condition has been implemented. For an incident beam approaching the rough surface from below, the computational region is doubled to include an "image ocean" above the real ocean. A complicated index of refraction is constructed in the image ocean (dependent on the rough surface) so that the image beam and the incident beam combine to give a zero total field on the rough surface. The computed field below the rough surface should then be the solution desired. The details of this method will be outlined. Examples will be presented showing that the split-step solution is identical to that obtained with a PE integral equation approach. Comparisons will also be made between the split-step and exact integral equation solutions, showing the effect of the PE approximation in rough surface scattering. [Work supported by ONR.]

4:36

QQQ14. Tests of Wiener–Hermite approximations for rough-surface scattering. Darrell R. Jackson and Eric I. Thorsos (Applied Physics Laboratory, College of Ocean and Fishery Sciences, Unversity of Washington, Seattle, WA 98195)

The Wiener–Hermite expansion has been used [Nakayama *et al.*, Radio Sci. **15**, 1049–1057 (1980)] to obtain a first-order approximation for wave scattering from randomly rough surfaces. Recently, a somewhat different Wiener–Hermite approximation has been developed [Meecham and Lin, J. Wave-Mater. Interaction **1**, 4–15 (1986)] and extended to second order [Meecham and Lin, IEEE J. Oceanic Eng., 357–361 (1987)]. This work is closely related to the smoothing method [Watson and Keller, J. Acoust. Soc. Am. **75**, 1705–1708 (1984)] and, like the latter, evidently includes multiple scattering effects while violating reciprocity. First-and second-order Wiener–Hermite approximations that satisfy reciprocity have been developed. The second-order approximation requires solution of an integral equation for the coherent reflection coefficient. This result is then inserted in an integral expression for the incoherent scattering cross section. Various Wiener–Hermite approximations have been tested by comparing them with exact numerical calculations. The numerical calculations were performed using a Monte-Carlo method [E. I. Thorsos, J. Acoust. Soc. Am. **83**, 78–92 (1988)], and the comparisons were restricted to the case of pressure release surfaces with Gaussian roughness spectra. [Work supported by ONR and ARO.]

4:48

QQQ15. Acoustic surface reflection loss at low grazing angles in the 0.95–1.35-kHz frequency band—A deep ocean measurement. Raymond J. Christian and Delia Klingbeil (Naval Underwater Systems Center, New London, CT 06320)

An experiment to measure acoustic reflection loss at the ocean surface was conducted in the North Sargasso Sea in December. A sonobuoy hydrophone received direct path and surface-reflected signals from a dipole element transmitter suspended from a research vessel. Both source and receiver had similar depths of approximately 80 m in the surface layer and separation ranges from 0.5–3 km. Pulsed linear frequency modulated transmissions from 0.95–1.35 kHz of 15-ms duration were used to perform matched filter correlations between the direct and surface reflected paths. Multipath separation was observed and surface reflection loss computations performed at specular grazing angles greater than 5 deg. Surface reflection loss versus grazing angle results under different sea-state conditions are used for comparison with those of the Bechman–Spezzichino scattering theory.

FRIDAY AFTERNOON, 18 NOVEMBER 1988 HONOLULU/KAHUKU ROOM, 3:15 TO 4:27 P.M.

Session RRR. Noise V: Aircraft and Aerospace Vehicle Noise in the Wilderness and the Community

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Contributed Papers

3:15

RRR1. Military aircraft overflight and wilderness perceptions. Paul H. Dunholter (Mestre Greve Associates, 280 Newport Center Drive, Suite 230, Newport Beach, CA 92660) and Robin T. Harrison (U. S. Forest Service, 444 East Bonita, San Dimas, CA 91773)

With the passage of Public Law 100-91, which mandates study by land management agencies of the impact of aircraft overflights on wilderness areas, has come increasing public scrutiny of military aircraft operations over such areas. Under the sponsorship of the U. S. Forest Service and the National Park Service, a preliminary study of such impact is being completed. Measurements of aircraft sound in the wilderness environment and assessment of park management perception of the effects of such sound indicate that such impact is not in itself adequately described by traditional metrics. The effect of sound onset rate on visitor reaction is also explored. Nonacoustic factors such as listener expectations and startle effect play an important role in visitor response to aircraft overflights. Indeed, the current work raises the question whether any objective acoustic measurement correlates with recorded visitor reaction.

3:27

RRR2. The impact of aircraft noise on elementary schools. Nelson D. Lewis and William A. Russell (Bio-Acoustics Division, U. S. Army Environmental Hygiene Agency, Aberdeen Proving Ground, MD 21010-5422)

The noise environment at elementary schools located under the flight paths at two military installations has been measured. These measurements showed that the day–night sound level (DNL) is a poor metric to use to describe this noise environment. The measured and predicted DNL's showed that these schools were located in an area described as acceptable for noise sensitive land uses. However, the measurements show that it will be difficult to maintain speech intelligibility in the classroom in excess of 25% of the time due to aircraft overflights.

3:39

RRR3. Hong Kong International Airport noise study. Areg Gharabegian (Engineering-Science, 75 N. Fair Oaks Avenue, Pasadena, CA 91109)

Existing noise problems associated with Hong Kong International Airport (HKIA) are severe. Noise levels from airport operations currently exceed government guidelines for at least a quarter of a million people. The measure used to date in Hong Kong to assess noise from airport operation is the noise exposure forecast (NEF). The baseline and future NEF contours were developed using the PC version of the integrated noise model (INM). Noise contours and flight tracks were generated by a computer-aided drafting (CAD) system using the INM data. Noise contour generated with CAD were superimposed on a digitized map of the Hong Kong planning areas to determine the population of each planning area that was affected by noise levels. Due to the unique landing pattern at HKIA, the INM noise profiles were calibrated for several major commercial aircraft. The future noise impact and recommended improvements will be discussed. Information will also be provided about using CAD and INM and calibrated noise profiles.

3:51

RRR4. The Valsan 727-200RE—A FAR part 36, stage 3, re-engining program. Alan H. Marsh (DyTec Engineering, Inc., 5092 Tasman Drive, Huntington Beach, CA 92649)

Boeing 727-200 airplanes powered by three Pratt & Whitney JT8D low-bypass ratio turbofan engines contribute much of the noise annoyance around most of the world's airports because of the high level of jet exhaust noise produced during departures and because of the high frequency of operations by the large number of 727-200 airplanes that are in service. Valsan Partners is managing a program to remove the two fuselage-mounted JT8D engines from 727-200 airplanes and replace them with newer higher-bypass-ratio JT8D-200 Series engines installed within new nacelles. A special internal-mixer nozzle is installed within a new acoustically treated tailpipe on the existing JT8D center engine. Re-engined 727-200RE airplanes produce significantly less noise during departures and also during landings. Cruise fuel consumption is also improved. Noise certification tests demonstrated compliance with the stage 3 noise level requirements of FAR part 36 as well as the chapter 3 requirements of ICAO Annex 16. The re-engining modification and test results will be described.

4:03

RRR5. The impact of transport category aircraft departure profiles on community noise exposure in the vicinity of John Wayne Airport, Orange County. Deborah W. Murphy (Douglas Aircraft Company, 3855 Lakewood Boulevard, Mail Code 36-60, Long Beach, CA 90846)

The noise resulting from transport category aircraft operations at John Wayne Airport, Orange County has confronted airport neighbors and airport operators and users since the introduction of turbojet aircraft to the airport in 1968. The unique distribution of population around the airport, coupled with the restricted number of commercial flights per day and the noise abatement flight procedures have led to a compromise which attempts to balance all airport activity with the community's needs for transportation and a livable environment. An analysis of numerous takeoff profiles was completed to determine the noise level sensitivity as a function of takeoff gross weight, aircraft configuration, and power reductions at various altitudes. Potential noise reductions can be obtained from the use of improved thrust and flap management techniques, tailored to the specific noise monitor requirements at John Wayne Airport, Orange County.

4:15

RRR6. Noise criteria methodology for site selection of the proposed Hokkaido Space Center in Japan. Marlund E. Hale (Engineering-Science, Inc., 75 North Fair Oaks, Pasadena, CA 91103) and Mori Ikegami (Shimizu Construction Co., Ltd., 261 South Figueroa Street, Suite 260, Los Angeles, CA 90012)

The U. S. space program has produced an enormous amount of data useful to environmental studies. Much of the launch noise data, however, is lacking correlation with such propagation influences as meteorological conditions and terrain effects. Thus, otherwise identical noise sources, such as the Space Shuttle, yield wide variations in measured and predicted launch noise. In addition, adequate criteria for assessing launch noise impacts remain an elusive issue within the several agencies and departments responsible for the evaluation of proposed new generation launch vehicles and possible locations for future space launch facilities. In order to develop a site selection criterion based on launch noise for the proposed Hokkaido Space Center, the limited acoustic, meteorological, and terrestrial information available from launches in Florida and California was evaluated, correlated, and scaled to designated launch vehicles and potential sites. The resulting launch noise site selection methodology was used in locating an acceptable site for both equatorial and polar orbit launches while providing consideration for facility support buildings and the surrounding region of potential acoustic impact.

Session SSS. Noise VI: Sound Diffraction, Vibration, and Community Noise Effects

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Sonoko Kuwano, Cochairman
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Contributed Papers

4:30

SSS1. Diffracted sound field behind a noise screen with horizontal slits. Zyun-iti Maekawa and Kouhei Tsuchiyama (Faculty of Engineering, Kobe University, Rokko-dai, Kobe, 657 Japan)

Highway noise screens can have some adverse effects on the human environment such as blocking sunshine, breeze, and view. In order to mitigate these adverse effects, the authors tried to create slits in such screens, and the effect of the slits on the noise reduction by the screen has been studied. Although the result for the vertical slits was reported at Inter-Noise 86, it has become clear that using horizontal slits is more convenient to satisfy the purpose mentioned above. A parametric study by calculation of the diffracted sound field using the line integral along the periphery of the screen under Kirchhoff's boundary condition was carried out with a personal computer. The results show that there are two zones behind the screen—that is, one increasing noise reduction and the other decreasing it—and that the shielding effect for highway noise is possible to be increased up to 7 dBA by the suitable design of a slitted screen.

4:42

SSS2. Finite element approaches to acoustic fields perturbed by a cylindrical tank. Tsutomu Hiramatsu, Yoshiaki Komura, and Masanori Shintani (Department of Mechanical Engineering, Fukui University, Fukui, 910 Japan)

The finite element method is applied to the acoustic field around a cylindrical tank in order to analyze the diffraction effects caused by the obstacle. A method used in this study is the so-called hybrid-type finite element method. The cylindrical tank situated in a finite domain perturbs the acoustic field radiated from a point source. The finite domain is divided into many finite elements, which are combined with the infinite elements so as to satisfy the continuity of the flux flow on the interelement boundary using an arbitrary interpolation function. The approach is compared with an experimental result which was obtained for the acoustic field around the water tank at a thermal power plant. From these studies, it can be seen that the hybrid technique using the combined finite and infinite elements is a useful method of analyzing and predicting the noise field around obstacles like water tanks.

4:54

SSS3. Community response to ground vibration excited by railway trains. Hisashi Hatanaka and Koichi Taya (Kobayasi Institute of Physical Research, 3-20-41 Higashimotomachi, Kokubunji, 185 Japan)

The relation between the vibration level excited by railway trains and community responses in the vicinity of railway tracks is discussed. The vibration levels excited by trains were measured at eight residential areas. In the same areas, questionnaires on residents' responses to their environment were obtained. The measured vibration levels were arranged into maximum value L_{max}, equivalent power value L_{eq}, and day and night average value L_{dn}. The relation between these arranged values and the community responses was analyzed by the method of multidimensional scaling, etc. The community response corresponded to the magnitude of all arranged values in the same order. However, the vibration level excited by freight trains had a higher correlation to the community response than that excited by electric trains.

5:06

SSS4. Cross-cultural study on noise problems in apartment houses in Japan, West Germany, the U. S. A., and China. Sonoko Kuwano, Seiichiro Namba (College of General Education, Osaka University, Toyonaka, 560 Japan), August Schick (Oldenburg University, D-2900 Oldenburg, West Germany), Mary Florentine (Northeastern University, Boston, MA 02115), and Zheng Da Rui (Academia Sinica, Beijing, People's Republic of China)

Neighborhood noise has become a serious problem in many countries, and it is difficult to regulate it by physical means alone. A cross-cultural survey was conducted in four countries, Japan, Germany, the United States, and China on the residents of apartment houses in order to examine the cultural differences involved. Interesting differences were found in preferred countermeasures, highly annoying sounds, attitudes against noise, expectations for laws, etc. The use of loudspeakers in public places was accepted in the four countries when they were used for conveying necessary information. The results suggest that it is important to take cultural or social backgrounds into consideration in order to find suitable countermeasures.

5:18

SSS5. Effects of low-level sound on the mental task of writing random numbers. Akihiro Tamura (Department of Architecture and Building Science, Faculty of Engineering, Yokohama National University, Yokohama, 240 Japan)

This study was undertaken to find out the basic problems involved in the effects of low-level sound on the performance of mental tasks by social psychological experiments in which subjects were not instructed that the test was to study the effects of sound on performance. The task was writing numbers from 0 to 9 randomly and successively. Subjects carried out the task under two different conditions; that is, a common condition and their own condition. They answered a questionnaire designed to investi-

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gate their feelings about the test room, the test condition, the experimenter, the co-experimenter, and themselves. These experiments were carried out three times for 3 years. They seem to justify the following conclusions. (1) The level of motivation has a major effect on the performance of the mental task. (2) Sounds which are not recognized as inevitable for behavior and the environment have a tendency to reduce performance even at low levels. (3) When the motivation for a task is present, low-level sounds, which are not recognized as inevitable, are found to be more annoying than those which are recognized as inevitable.

5:30

SSS6. Noisiness of impulsive sound in background noise. Hiromitsu Miyazono, Keishi Ohto, Tsuyoshi Usagawa, and Masanao Ebata (Faculty of Engineering, Kumamoto University, 2-39-1 Kurokami, Kumamoto, 860 Japan)

There are many studies of rating noise, but the method has not been established completely yet. Especially in the rating of impulsive noise, some problems remain unsolved even though an evaluation based on energy level is effective to some extent. In this study, the noisiness of impulsive sound is measured when background noise is present at various levels. The results show that the noisiness decreases as background noise increases and can be almost completely evaluated by the difference in the energy level of the impulsive sound and the background noise. However, when the level of the background noise is comparable to that of the impulsive sound, there is a considerable difference between the estimated value and the observed noisiness. To derive an accurate rating, two alternative methods are discussed: one is based on NOY and the other based on a background correction which is similar to a loudness rating.

5:42

SSS7. The ultimate community noise monitoring system. Ellwyn G. Brickson (Health Care Agency, Box 355 Building 50, Santa Ana, CA 92702)

Community noise measurements of intruding noise sources in Orange County have usually been conducted with hand-held or integrating sound-level metering systems. The ambient noise is measured either before or after the test period. When the intruding noise cannot be turned off or vehicular traffic is a major contributor to the problem, the noise must be controlled to meet a community noise standard. A more sophisticated system must be utilized to achieve compliance during an outdoor concert. One system proposed for Pacific Amphitheatre concerts would use a real time analyzer to transmit bursts of octave band pink noise from the speaker system prior to each concert. The signal would be monitored at three locations in the residental area to determine attenuation and make the correct assessment of such intruding noise. A lap-top computer can then be used to assess the noise dose and to prescribe the amphitheatre music sound level reduction so as not to exceed noise ordinance standards.

FRIDAY AFTERNOON, 18 NOVEMBER 1988 KAUAI ROOM (WEST END), 2:00 TO 5:00 P.M.

Session TTT. Late Papers (Poster Session)

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A list of late papers will be posted in the Society Registration Area